

DIGITAL FILTERING TECHNIQUES FOR PROCESSING
SIGNALS UTILIZED IN THE UNMANNED SUPERVISION OF MACHINING

By

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To Catherine and Anastasia for their beautiful inspiration;

A bed time story sure to put anyone to sleep.

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The application of digital filtering techniques is applied to the task of unmanned supervision of machining. Unmanned supervision of machining is desirable to free the operator from constantly overseeing the machining process and to expand his capability to detect in-process cutter breakage and control chatter. The filtering task is presented in two areas; the filtering of a time synchronous sound signal used in the detection of chatter, and filtering a rotation synchronous force or displacement signal in the detection of in-process cutter breakage.

In the area of filtering for the detection of chatter, the use and limitations of a revolution referenced finite impulse response comb filter is investigated. The issue of filter notch misplacement due to filtering the revolution synchronous signal utilizing a time based sampling is addressed, and system design parameters are established for a

frequency range of interest versus sampling frequency. The performance of the filter during transient spindle speeds is also presented. Two alternative filtering techniques are investigated; an infinite impulse response comb and a high low pass comb filtering technique.

In the area of in-process cutter breakage, two filtering sub systems are presented. The first is based on a once per revolution difference comb filter to detect catastrophic in process breakage. The sub system utilizes a moving average filter to suppress signals during transient milling and has automatic thresholding. The second sub system is utilized to detect slowly progressing cutter damage and the existence of previous damage to the cutter. The sub system is designed to be insensitive to cutter run-out and relies on a calibration cut to establish threshold levels.

CHAPTER 1 INTRODUCTION

The purpose of this dissertation is to demonstrate the application of digital filtering techniques to the unmanned supervision of machining. Filters are applied to two different areas of specific interest; filtering of a sound signal for use in the detection and correction of chatter, and to the detection of milling cutter breakage in process. It is of interest to automate these systems for several reasons. The implementation of unmanned milling frees the operator from simply watching the CNC machine during the cutting process, this liberates the operator's time, allowing one operator to operate several machines at once and relieves him of tedious supervision tasks. In addition, the automated system is designed to increase the abilities of the operator to detect and correct machining problems that were previously beyond his capabilities.

Filtering is applied in the detection and correction of chatter during machining operations. Chatter is the unstable self excited vibration of the cutting tool relative to the work piece. Operation of the machine in a chattering condition leads to unacceptable finish, out of tolerance parts and possible damage to the tool or machine. The mechanism of chatter is controlled by the dynamic stiffness of the machine-tool-work piece system. It has been shown by Tlustý [1] that chatter is strongly dependent on the phasing of the waviness of the surface with the position of the

milling cutter teeth. This leads directly to chatter stability being strongly dependent on the spindle speed of the cutting machine. The stability of the operation exhibits lobes of stability versus the spindle speed. Because of this, it is of interest to operate the machine in one of its highly stable lobes thus reducing the instability and allowing higher stable metal removal rates. An automated control system to do this has been developed by Tlustý, Smith and Delio [2,3,4]. The system known as the chatter recognition and control, CRAC, utilizes a signal from the cutting process that contains frequency information from the chattering cut. This signal may be a force signal, a displacement signal, or any other representative input. In its present form the CRAC system utilizes a sound signal from the cutting process. This signal is used as a matter of convenience for instrumenting the process. As the sound signal contains information from the vibration of both the tool-machine system and the work piece, it can be used to detect chatter of either the tool or the work piece. The corrective action for both these conditions is the same. Further, the microphone is not sensitive to its placement depending on the mode of vibration; for instance, if an accelerometer is used it must be located away from the nodes of vibration in order to sense the chatter. As the mode of vibration is not previously known, the placement of the accelerometer can not be a priori determined. Further, placement of the accelerometer on the machine structure may completely isolate it from chatter of the work piece.

The chatter recognition and control system is designed to automatically recognize chatter and direct the cutting machine to its stable speeds. To do this automatic recognition of chatter, the system must be able to distinguish between the

signal of a stable cut and that of the chattering cut. The signal from the cutting process is made up of many components. Some are normal content of the signal from a stable cut. There are also noise components due to various sources such as electrical equipment and bearing or transmission sources and the signal component due to the chattering of the process. The chatter control system recognizes chatter as a spectral peak that exceeds a predetermined background threshold that is not due to normal cutting signals or known noise sources. It is thus the task of the filtering system to remove the content due to normal cutting and all known noise sources from the signal so that the only content remaining will be that of the chatter. Therefore any spectral peak in the filtered signal that exceeds the threshold is determined to be that of chatter and the control system acts on this signal.

By recognizing and controlling chatter, the system allows the operator to correct a chattering condition, which would not be possible without the monitoring system. Although an experienced operator can recognize chatter by its distinctive sound and the surface left by the cutter, it is not possible to properly correct the cutting conditions without knowing the frequency content of the cutting signal. Hence, the use of this system not only allows less manned supervision of the cutting process, it allows enhanced performance that would not be possible without the system.

Filtering is also applied to the detection of milling cutter breakage. It is desirable to detect milling cutter breakage in process. If the cutter edges break during the cutting process, the surface produced can deteriorate giving an unacceptable finish. Further, during heavy cutting when the cutter edges are highly loaded, the breakage of

a tooth can overload the subsequent teeth, leading to rapid failure of the remaining teeth. This can produce damage or destroy the cutter, work piece or machine tool spindle itself. Thus it is important to quickly detect the breakage of a cutting edge and stop the cutting process before damage can spread. The signal due to tool breakage must also be definitively distinguished from other signals due to normal cutting operations. This must be done to avoid false alarms from the detection system. False alarms lead to stoppage of the cutting process and inefficiencies associated with these stoppages. Further, if the system consistently issues false alarms it will become recognized as unreliable and will be disconnected and unused.

The system to detect cutter breakage utilizes a force or displacement signal in order to detect breakage. The displacement signal is that of the forced displacement of the spindle relative to the machine tool frame. The displacement signal is used to represent the force on the cutter. It must be understood that this is not the true force signal, but the force signal filtered through the displacement versus force transfer function of the individual tool and spindle combination. The signal used in the cutter breakage detection system is synchronously sampled; that is the signal is sampled in synchronization with the spindle rotation. Each sample is acquired at the same position during the spindle rotation on each revolution. This synchronizing leads to great simplicity in the filtering of the signal and non-varying coefficients for digital filters that are applied to it.

In the detection of cutter breakage it is, as in chatter, necessary to separate the signal content due to normal cutting conditions from those due to the event that is

sought. The task of the filtering is to remove the signal content due to normal operation from those signals that are indicative of cutter breakage. The other major task of the cutter breakage system is to establish a threshold which delineates between breakage and nonbreakage of the cutter. This threshold varies based on parameters of the cut, such as the radial immersion. The overall system is based on two sub-systems. One is designed to eliminate signal content due to cutter run-out and react to in process breakage. This sub-system is insensitive to slow damage such as chipping or beginning the cutting process with a previously damaged cutter. The second sub-system is slower reacting but is sensitive to the absolute condition of the cutter. That is, the system is sensitive to slowly developing cutter damage and previously damaged cutters.

The cutter breakage system frees the operator from watching and listening for damaged cutters. With the system in place it is practical to push the cutter at higher loads and longer in to the cutter life. If the system is reliable, the cutter can be used more aggressively without the danger of a tooth failure destroying the cutter or work piece.

CHAPTER 2

THE FILTERING TASK

Digital filters in chatter control are used to remove signal content due to normal cutting and known noise sources from the content due to chatter in the cutting process. Machine tool chatter is a self excited vibration of the tool relative to the work piece. This self excited vibration occurs at the most dynamically flexible mode of vibration of the system, provided steps have not been taken to stabilize a mode. The task of chatter recognition and control is to recognize the signal due to chatter and adjust the machining conditions to the most stable configuration. In order to recognize chatter, the signal content due to normal cutting and known noise sources must be filtered from the cutting signal. In this way the remaining content is due to chatter and the control system can act on this signal.

In the present chatter control system at the Machine Tool Research Center at the University of Florida, a microphone is used to sense chatter. The microphone is nonintrusive and senses both vibration of the tool and of the work piece. A full description of the chatter control system can be found in the works by Delio, Smith and Tlusty [2,3,4]. The signal gathered contains content due to the normal cutting, noise from other sources such as electrical or bearing passing, and the signal from the chatter vibration if it is present. The signal content due to normal cutting conditions comes from the periodic striking of the cutter teeth with the work piece.

Because this periodic striking is not harmonic in nature, in fact it may resemble a series of pulses, the signal contains frequency content at the tooth striking frequency and its harmonics. It can be shown that the regenerative feedback mechanism of variable chip thickness that causes the instability of chatter is defeated by operating the machine such that the tooth frequency is equal to the natural frequency of the mode of chatter of the machine [1]. Further, it can be shown that the mechanism is defeated if there is an integer number of vibration waves between each successive tooth of the milling cutter. In this way the phase of vibration does not change from tooth to tooth. This constant phase produces a constant thickness chip, thus causing no regeneration. Because of this, chatter cannot occur at the tooth frequency or its harmonics. Therefore, it is always desirable to filter the tooth frequency and its harmonics from the signal collected to sense chatter.

There is also content in the milling signal due to slight tooth throws. A tooth throw is the misplacement of a cutting tooth off the ideal circle of the cutting tool. This is shown in Figure 2-1. The misplaced tooth inputs a once per revolution pulse

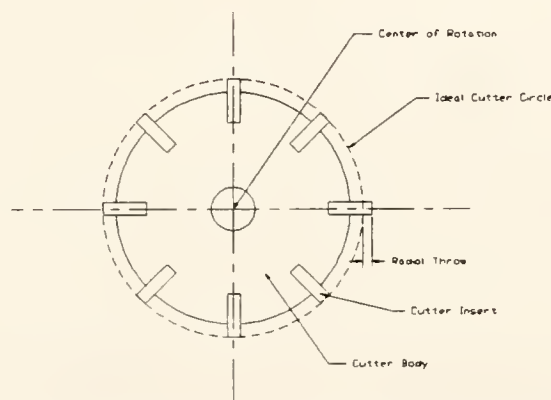


Figure 2-1 Radial Throw Diagram for an Inserted Milling Cutter.

into the cutting signal. The duration of this pulse is dictated by the radial immersion of the cutter. The once per revolution pulse contributes frequency content at the once per revolution frequency known as the run-out frequency and its harmonics. The shorter the pulse, that is the lower the radial immersion, the farther the power will be spread into the harmonics. This signal content must also be filtered from the chatter signal. The run-out harmonics are not stable milling conditions as was the case with the tooth frequency harmonics. Therefore, chatter can occur at these frequencies. It is thus desirable to remove only those run-out harmonics necessary to avoid false triggering of the chatter detection system and leave the remaining harmonics unfiltered to avoid filtering of a valid chatter signal.

Other signals that must be filtered from the chatter signal can be separated into two classes. Frequency stationary signals, such as electrical noise, remain at a set frequency no matter the operational conditions of the machine tool. There is also noise that varies in frequency but is located at a set multiple of the spindle rotation frequency. Signals such as this arise from various sources such as air passing over a bearing or the noise from a gear train attached to the spindle. These signals must be removed from the chatter detection signal to avoid false triggering.

A short introduction to digital filtering basics is included here. This is not meant to be a comprehensive introduction, but only to introduce those concepts necessary in understanding the filters implemented and the reasons behind them. A complete introduction to digital filtering and general digital signal processing is

included in Strum and Kirk, Discrete Systems and Digital Signal Processing [5].

Much of this basic introduction is condensed from this text.

Digital signal processing is based on the processing of digitally sampled signals. These signals are commonly sampled at equal time intervals, the time interval is dictated by the sampling frequency, f_s . The digital frequency, Θ , of a signal is defined as the ratio of the frequency of the signal to the sampling frequency times two times pi.

$$\Theta = 2 * \pi * \frac{f}{f_s}$$

By utilizing the digital frequency of a signal the individual sampled values can be attained by simply indexing the sample number. For example, the sampled values of a cosine wave of frequency, f , sampled at a frequency f_s can be evaluated from the following.

$$\cos(\Theta * n) \quad n=0, 1, 2, 3, \dots$$

It is obvious that the Nyquist frequency, $f_s / 2$, (the maximum frequency that can be resolved) has a digital frequency of π .

A simple representation of a digital filter is shown in Figure 2-2. The input of the filter is represented by the series of samples $X(n)$ and the output is represented by $Y(n)$. Digital filters can conveniently be separated into two classes; finite impulse filters (FIR filters) and infinite impulse filters (IIR filters). For a FIR filter the output of the filter depends only on the input, while the output of an IIR filter depends both on the input series and past values of the output series.



Figure 2-2 Filter Diagram

The output of a FIR filter can be written as linear combination of the input series. Thus, $Y(n)$ can be represented as follows:

$$Y(n) = b_0 X(n) + b_1 X(n-1) + \dots + b_k X(n-k) = \sum_{k=0}^k b_k X(n-k)$$

where k is the order of the filter and the b 's are the filter coefficients. This series is commonly represented by the flow diagram shown as Figure 2-3, where the D represents a delay of one sample period and the b 's are again the filter coefficients. If a delay operator, Z , is defined the series can be written in terms of this delay operator as follows:

$$Y(z) = (b_0 + b_1 z^{-1} + b_2 z^{-2} + \dots + b_k z^{-k}) X(z)$$

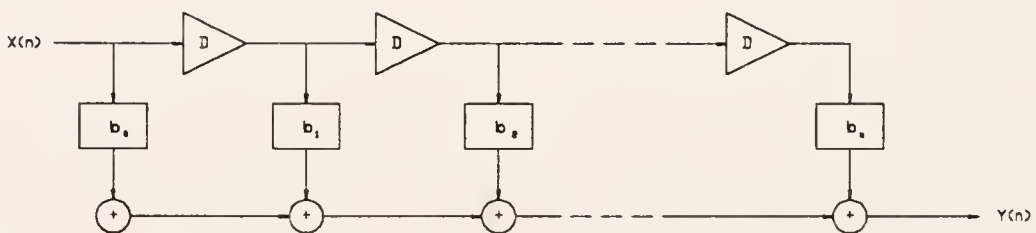


Figure 2-3 Signal Flow Diagram for a Finite Impulse Response Filter

The transfer function of the filter, $H(z)$, can be written as the output over the input.

$$H(z) = \frac{Y(z)}{X(z)} = \frac{(b_0 + b_1 z^{-1} + b_2 z^{-2} + \dots + b_k z^{-k})}{1}$$

The roots of the Z polynomial in the numerator of the transfer function are known as the 'zeros' of the transfer function. It can be shown that for real input and real output, these roots will either be real or occur in complex conjugate pairs. The rooted transfer function can be written as follows.

$$H(z) = \frac{(z - R_1 e^{j\Theta_1}) (z - R_1 e^{-j\Theta_1}) \dots (z - R_{\frac{k}{2}} e^{-j\Theta_{\frac{k}{2}}})}{1}$$

These roots can then be plotted in the complex Z plane as in Figure 2-4. Only a single pair of complex conjugate roots are shown in the figure. The circle in the figure is the unit circle of radius one. The frequency response of the filter transfer function may be evaluated by replacing the delay operator Z with $e^{j\Theta}$, where Θ is the digital frequency at which the response is to be evaluated. By varying theta continuously from zero to pi, the continuous Fourier transform of the impulse response of the filter may be evaluated. By evaluating the function at discrete locations with spacing equal to two pi divided by the number of points in the transform, the discrete fourier transform is evaluated. The section of the transform past the Nyquist frequency of pi will be a mirror image of the section below pi, thus it is only necessary to evaluate the function from zero to pi.

The evaluation of the frequency response can be thought of in graphical terms by imagining what is happening in the complex Z plane. Figure 2-5 shows a single complex pair of roots with radius equal to one in the complex Z plane. The transfer

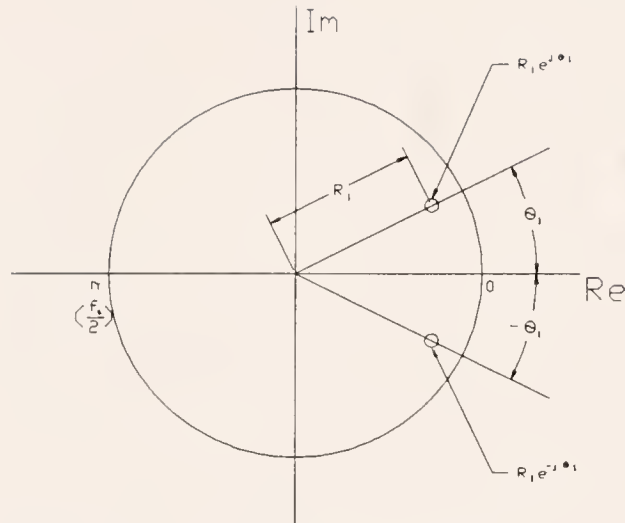


Figure 2-4 Pole-Zero Plot for a Second Order FIR Filter.

function can be written as

$$H(z) = \frac{(z - 1e^{j\theta_1})(z - 1e^{-j\theta_1})}{1}$$

The frequency response is evaluated by replacing Z with a unit vector at the desired frequency, $e^{j\omega}$. The magnitude of the frequency response is then the product of the magnitude of each the numerator terms. The magnitude of a numerator term is the length of the vector from the tip of the $e^{j\omega}$ vector to the root for that term. This is represented in the figure as N_1 and N_2 for the two numerator roots. The overall magnitude can then be written.

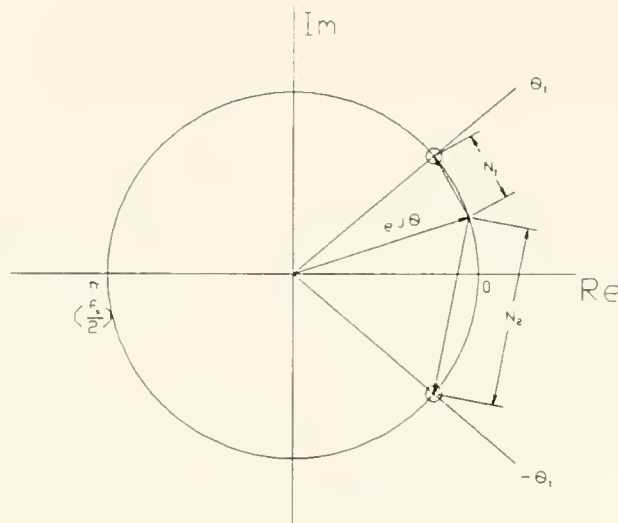


Figure 2-5 Graphical Magnitude Evaluation for a FIR Filter Transfer Function.

$$|H(e^{j\theta})| = \frac{N_1 N_2}{1}$$

The complete frequency response can be pictured this way. The magnitude of the frequency response from zero to pi is plotted as Figure 2-6. For low frequencies near theta equal zero, both N_1 and N_2 will be less than one, thus the overall magnitude is less than one and signals in this frequency range are attenuated. As theta approaches theta₁, N_1 becomes very small and this term dominates the overall response of the filter. Because the root is on the unit circle, when theta equals theta₁, N_1 goes to zero. The filter completely removes any signal content at this frequency. There is a zero in the magnitude, hence the term zero for the root. As the frequency is further increased, both N_1 and N_2 become larger than one and the filter amplifies signals at the higher frequencies.

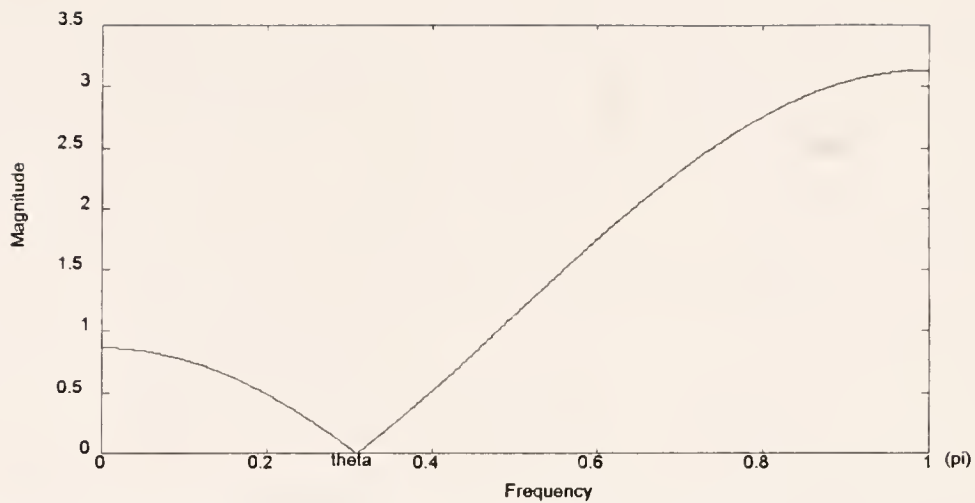


Figure 2-6 Magnitude Response of a Second Order FIR Filter.

Infinite impulse response filters depend both on the input and past output of the filter to form the output series. The filtering action can be written as follows:

$$y(n) + a_1 y(n-1) + \dots + a_m y(n-m) = b_0 x(n) + b_1 x(n-1) + \dots + b_k x(n-k)$$

where the terms are as previously mentioned and the a's represent the coefficients placed on the past output values of the filter. This can be more compactly written as follows:

$$y(n) = \sum_{k=0}^k b_k x(n-k) - \sum_{m=1}^m a_m y(n-m)$$

A flow type diagram for an IIR filter is shown in Figure 2-7. There is the possibility of an unstable filter. The transfer function of the filter can be written in terms of the delay operator Z as follows:

$$H(z) = \frac{Y(z)}{X(z)} = \frac{(b_0 + b_1 z^{-1} + b_2 z^{-2} + \dots + b_k z^{-k})}{(1 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_m z^{-m})}$$

The roots of the denominator polynomial are known as the 'poles' of the filter.

The poles must lie within the unit circle in the Z plane in order for the filter to be stable.

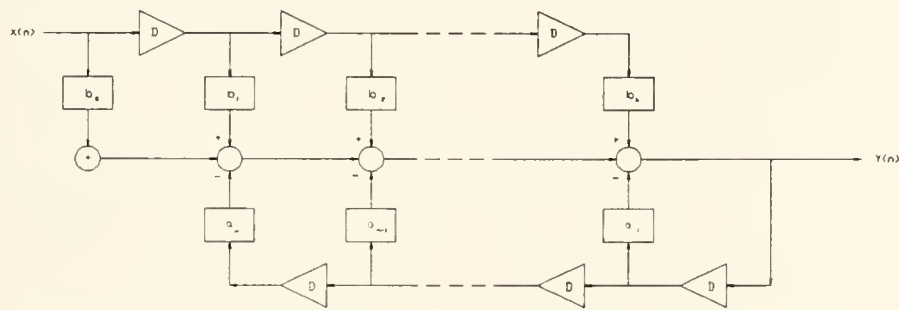


Figure 2-7 Signal Flow Diagram for an Infinite Impulse Response Filter.

Just as with the FIR filter, the filter response for the IIR filter may be pictured in the Z plane. A diagram for a two pole, two zero filter is shown as Figure 2-8. The zeros are again located on the unit circle and the poles are located at the same digital frequency, Θ_1 , as the zeros but inset from the unit circle. This is of course not a requirement but used for illustration of the magnitude evaluation. The transfer function can be written in its rooted form as follows:

$$H(z) = \frac{(z - 1e^{j\Theta_1})(z - 1e^{-j\Theta_1})}{(z - R_1e^{j\Theta_1})(z - R_1e^{-j\Theta_1})}$$

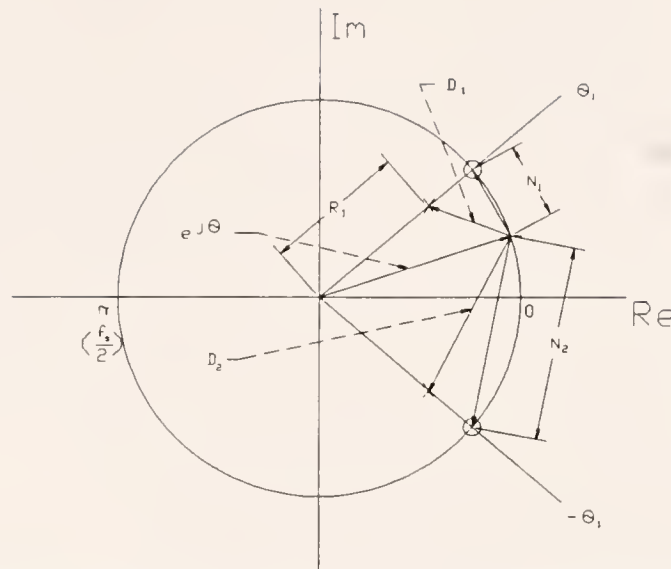


Figure 2-8 Graphical Magnitude Evaluation for an IIR Filter Transfer Function.

The magnitude is again the product of the magnitude of the individual terms of the polynomials, with the numerator product now divided by the denominator product instead of one.

$$|H(e^{j\theta})| = \frac{N_1 N_2}{D_1 D_2}$$

The terms D_1 and D_2 are the magnitudes of the vectors from the tip of the $e^{j\theta}$ to the poles. The magnitude response is shown as Figure 9, as the frequency approaches the frequency of the zero, the magnitude still goes to zero as was the case with the FIR filter. But, because the vector D_1 is also shrinking, the magnitude is less effected until the frequency closely approaches the zero. This results in a much sharper or higher quality notch in the frequency response function.

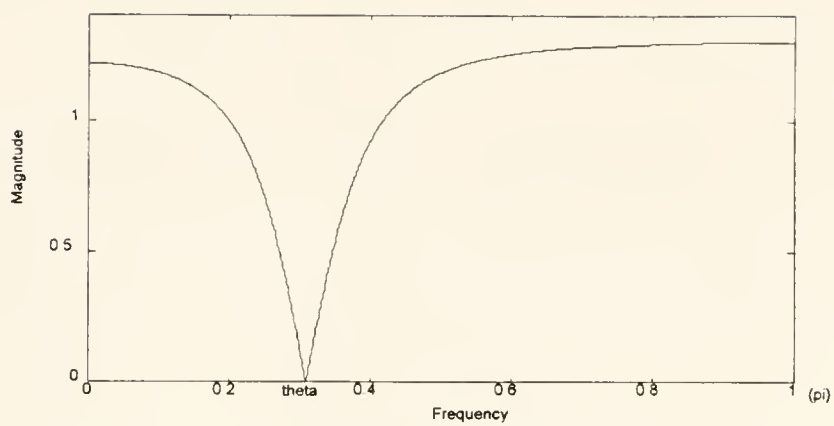


Figure 2-9 Magnitude Response of a Second Order IIR Filter. Pole Radius 0.75.

CHAPTER 3 REVOLUTION REFERENCED COMB

In its present form, the chatter recognition and control system utilizes a specialized form of a FIR comb filter. This filtering method has been described by Tlusty et al. [6] and by Frost [7]. The comb filter is a filter with its zeros equally spaced around the unit circle. This equal placement of the zeros leads to a very simple difference equation used to implement the filter. In general a filter with k zeros would have k terms in the rooted numerator. This leads to a numerator polynomial with k+1 coefficients. In order to implement this, the coefficients must be calculated and then each multiplied by the appropriate sample in the input series and then summed to form the out put sample. Neglecting the evaluation of the coefficients, this leads to k+1 multiplies and k additions for each out put sample. For the comb filter with the zeros located on the unit circle, the terms of the factored numerator can be written

$$(z - 1 e^{j(\frac{m2\pi}{k})}) \quad m=0, 1, 2, 3 \dots k-1$$

The product of these terms leads to the simple form of the comb filter transfer function.

$$H(z) = \frac{1 - z^{-k}}{1}$$

It is seen that in order to implement the k^{th} order comb filter, only a single subtraction of the sample k delays back, from the present sample, is required to form the out put sample.

The action of the comb filter is easily understood in the time domain. If the filter is applied to a signal from a rotating cutting tool, and an integer number of samples is collected each revolution, by subtracting the sample one revolution back from the present sample any signal that repeats each revolution will be canceled. This is represented by the simple difference equation used to implement the filter.

$$Y(n) = X(n) - X(n-k)$$

There will be a zero at the run-out frequency and each of its harmonics. The delay number or filter order is equal to the number of samples taken each revolution.

The zero plot of a 15^{th} order comb filter is shown as Figure 3-1. The associated magnitude response of the filter is shown in Figure 3-2. It can be seen that the equally spaced zeros produce equally spaced notches in the magnitude of the filter transfer function. Also the signal between these zeros is distorted. If a signal is exactly in between two zeros it will be amplified by a factor of two. This is due to the subtracted wave at these points being out of phase with the present signal. In this way the subtraction adds the two waves producing constructive interference. It can be shown that the actual form of the magnitude can be written as follows.

$$|H(z)| = |2\sin(k\theta)|$$

Therefore any signal content falling in the range about a zero of

$$\theta_{zero} - \frac{\pi}{6k} < \theta < \theta_{zero} + \frac{\pi}{6k}$$

will be attenuated by the filter.

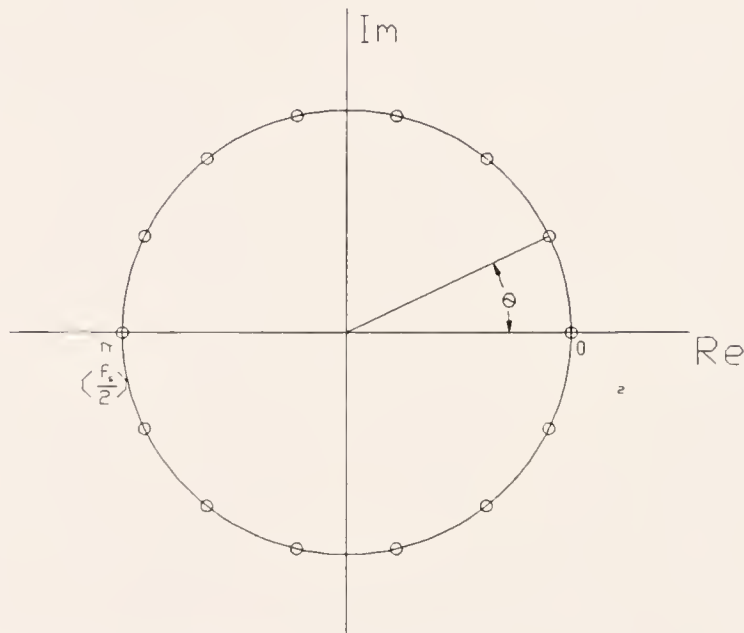


Figure 3-1 Zero Plot of a 15th Order Comb Filter.

The actual implementation of the comb filter is done using two signals. This system is referred to as the Revolution Referenced Comb. The two signals used are the sound signal from the cutting process and a once per revolution signal from the spindle. The sound signal is of course the signal that is to be filtered for the detection of chatter. The once per revolution signal is used to adjust the filter. Because the sound signal is sampled at a set sampling frequency, there is not the same number of samples per revolution of the spindle for different spindle speeds. Therefore, the delay number used in the comb filter must be adjusted to match the number of samples per

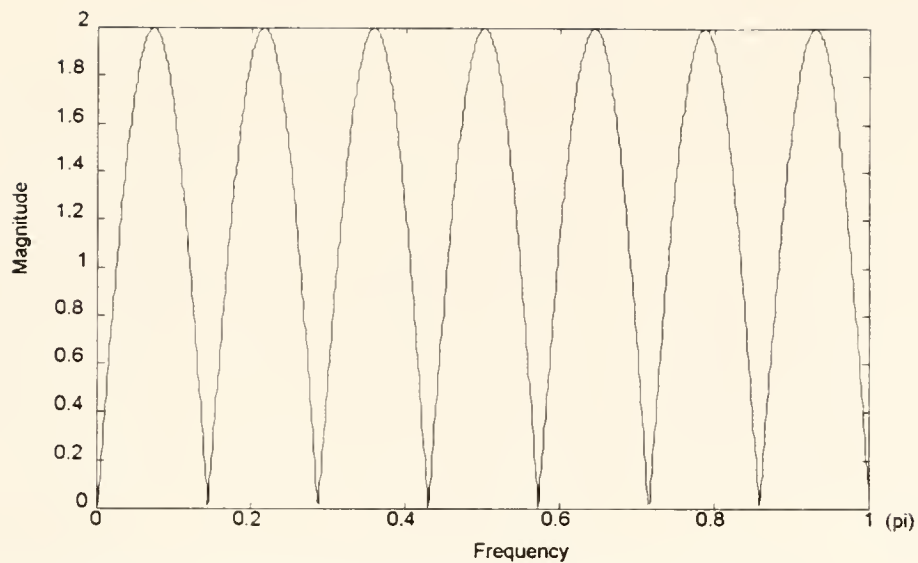


Figure 3-2 Magnitude Transfer Function of 15th Order Comb Filter.

revolution at any given speed. This is accomplished by using the once per revolution signal from the spindle. The number of samples acquired between successive once per revolution signals is counted by the controlling program. In this way the number of samples per revolution is known and hence the proper delay for the filter is known.

Due to the method of the setting of the filter delay and the signal being sampled at a constant frequency, there is a misplacement of the filter notches in this method. Because the spindle speed is in no way synchronized with the sampling of the data, there is not normally an integer number of samples per revolution of the spindle. However, the comb filter must operate with an integer delay number (a non integer delay would require interpolation between the sampled values). This means that the true delay must be rounded to the nearest integer. In fact, due to the method of counting the number of samples, the delay is always rounded down to the integer value.

To count the delay number the controlling program counts the number of samples acquired between edges of the once per revolution signal. If the once per revolution edge comes just previous to a sample being taken, the sample period was indeed in that revolution. However, it would not be counted by the system as the sample was not acquired in that revolution. By this mechanism the delay is always rounded down to the integer value. It can be seen that in the limit case the counted delay would be one sample period shorter than the actual revolution of the spindle. The digital frequency of the run-out signal can be calculated as follows:

$$\Theta_{ro} = 2\pi \frac{f_{ro} \left(\frac{rev}{sec} \right)}{f_s \left(\frac{samples}{sec} \right)} = \frac{2\pi}{N}$$

where N is the number of samples per revolution. The maximum misplacement of the filter notches can then be calculated

$$\Theta_{err} = \Theta_{true} - \Theta$$

where Θ_{true} is the true digital frequency, Θ is the calculated one from the once per revolution signal.

$$\Theta_{err} = \frac{2\pi}{N_{true}} - \frac{2\pi}{N}$$

The true noninteger number of sample periods in the revolution is N_{true} and N is the counted number of samples per revolution. By the limiting condition imposed by the counting system,

$$N_{true} = N + 1$$

therefore the maximum misplacement can be calculated

$$\Theta_{err} = - \frac{2\pi}{N^2 + N}$$

It is seen that the misplacement of the filter notches is strongly dependent on the number of samples per revolution, that is, the ratio of the spindle run-out frequency to the sampling frequency. Obviously if there are many samples on a wave the misplacement of the subtracted wave by a sample period will have little effect, while if only a few samples are available the misplacement by one represents a significant phase shift and will greatly disturb the filtering action. Further, Θ_{err} is the misplacement of the primary notch at the run-out frequency. The notches at the higher harmonics are misplaced by a multiple of this error. This is due to having fewer and fewer samples on each wave as the frequency increases.

From the previous discussion it is seen that in order to diminish the misplacement of the filter notches, the highest possible sampling frequency should be used. The data must be sampled at this high frequency, it is not sufficient to sample the once per revolution signal at a higher frequency to better resolve the spindle speed. Unless some sort of interpolation scheme is used, the comb filter must work on integer numbers of delays therefore requiring the high speed sampling of the data.

For a set sampling frequency and a set range of interest in the frequency response of the machine tool, the misplacement of the filter notches is independent of spindle speed. For example, if the spindle speed is such that there are ten harmonics of the run-out in the frequency range of interest (referring to the primary signal as the

first), then the misplacement of the notch for the tenth harmonic is $10 \cdot \Theta_{err}$. If the spindle speed is increased to ten times its original speed, the misplacement of the primary notch is now ten times the misplacement of the original primary notch, and therefore has the same misplacement as the previous higher harmonic notch. This may not give the exact same effect in practice, because the power in the primary and the harmonics are not necessarily equal.

Filter Notch Misplacement

The performance of the revolution referenced comb is presented with emphasis on two areas of concern, the effect of notch misplacement and the effect of speed variation on the filtering. The effect of notch misplacement is investigated through the use of simulated cutting signals. Simulated cutting force signals are shown in Figures 3-3 A and B. The signal in A is from a four tooth cutter operating at 2500 RPM and a sampling frequency of 10,000 Hz. This gives exactly 240 samples per revolution, therefore there is no misplacement of the filter notches. The signal in B is from a similar cutter operating at 2490 RPM. This gives 240.9 samples per revolution. In both cases the revolution referenced filter would filter the signal with a delay count of 240. This produces a misplacement of the primary run-out notch of 97.8×10^{-6} radians in case B (the maximum possible for a 240 sample delay being 108.6×10^{-6} radians). Therefore case B has nearly the maximum misplacement of the filter notches that can be. Both simulated cuts are run with very shallow radial immersion, five percent, to assure that the tooth harmonics extend throughout the spectrum. A

tooth throw of ten percent of the feed per tooth is included to introduce the runout harmonics into the spectrum. The x direction force is plotted.

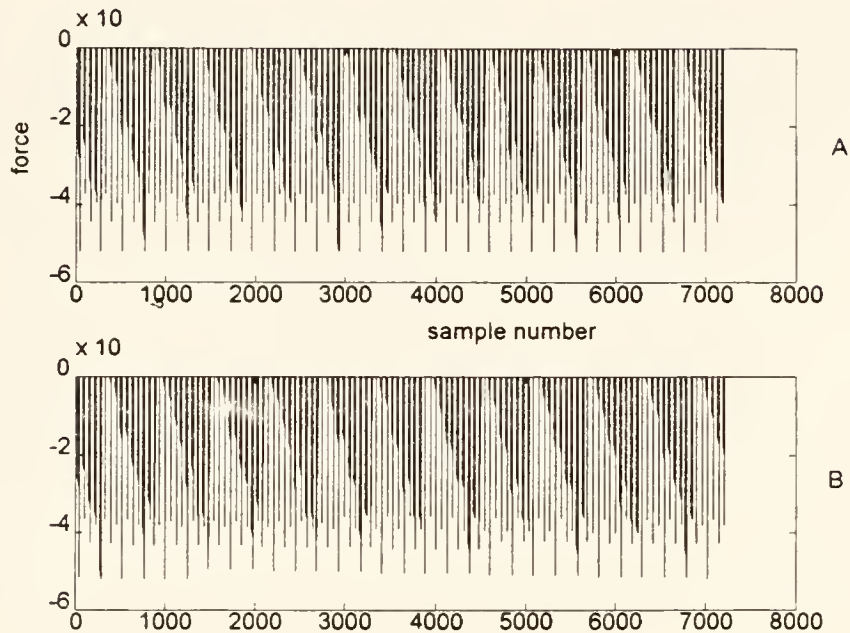


Figure 3-3 Simulated X Direction Force Signals for a Four Flute Cutter with 5% Radial Immersion and 10% Tooth Throw, Sampling Frequency 10,000 Hz. A) 2500 RPM B) 2490 RPM.

The spectrum of the 2500 RPM cut is shown in Figure 3-4. The spectrum shows strong content at the tooth harmonics as would be expected for the shallow immersion cut and minor content at the run-out harmonics due to the tooth throw. The spectrum of the signal filtered through a 240 delay comb filter is shown in Figure 3-5. The filter almost completely removes all the content of the signal. This is expected as there is no notch misplacement for the case.

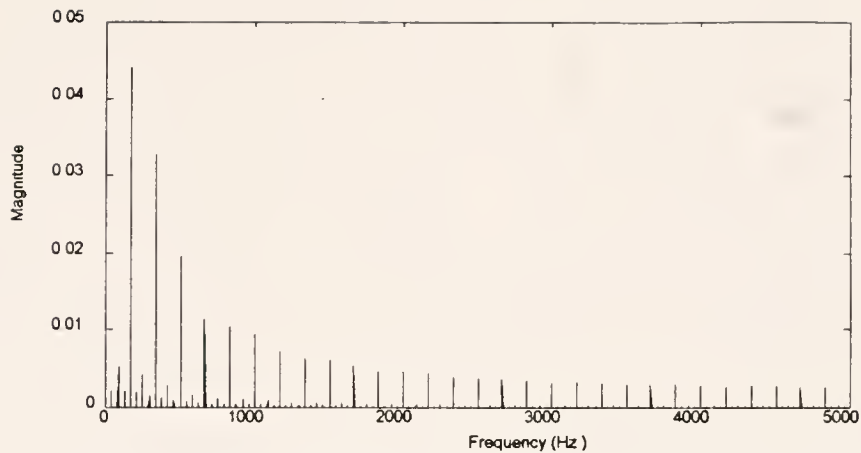


Figure 3-4 Spectrum of Four Tooth 5% Radial Immersion Cut, 2500 RPM, Sampling Frequency 10,000 Hz.

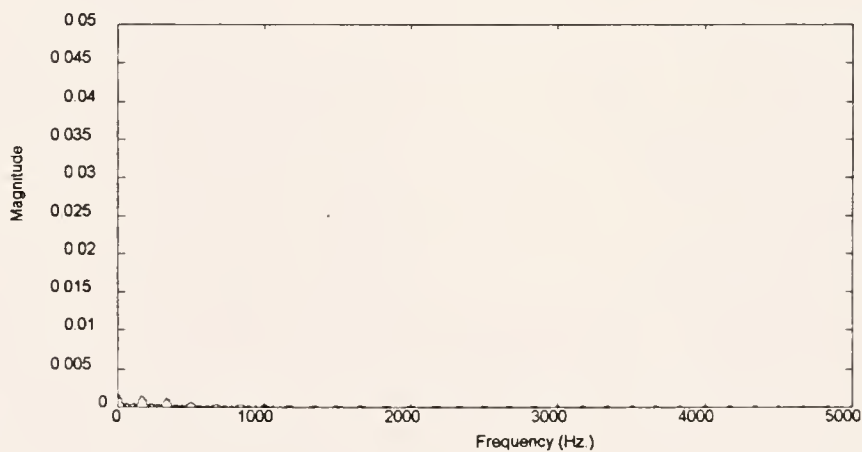


Figure 3-5 Spectrum for Four Tooth 5% Radial Immersion Cut, 2500 RPM, Sampling Frequency 10,000 RPM, Filter through a 240 Delay FIR Comb Filter.

Figure 3-6 shows the spectrum of the 2490 RPM cut. The spectrum is essentially the same as that of the 2500 RPM cut except for a slight frequency shift due to the different spindle speed. The signal is then filtered through the 240 delay comb and the spectrum shown in Figure 3-7. The filtered spectrum is plotted with the same scale as the unfiltered to allow direct comparison of the magnitude. It can be

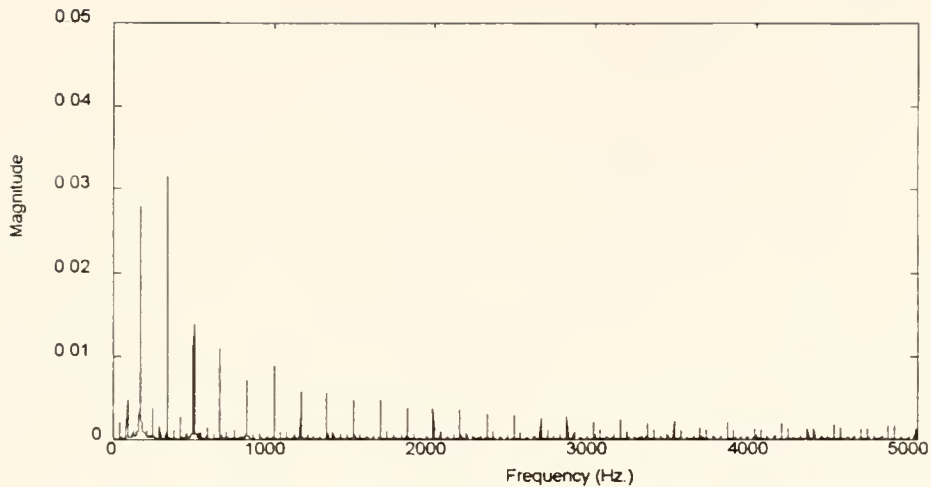


Figure 3-6 Spectrum of Four Tooth 5% Radial Immersion Cut, 2490 RPM, Sampling Frequency 10,000 Hz.

seen that the misplacement of the filter notches has caused the filter to leave a significant portion of the signal. Further, past a certain frequency in the spectrum the harmonic lines are not filtered out but in fact amplified by the revolution referenced filter. The performance of the filter at this limit case with maximum notch

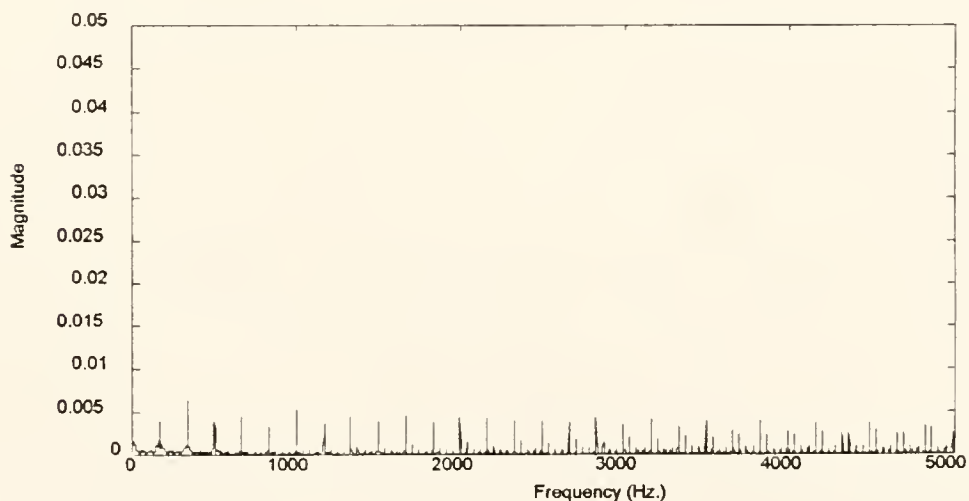


Figure 3-7 Spectrum of Four Tooth 5% Radial Immersion Cut, 2490 RPM, Sampling Frequency 10,000 Hz. Filtered Through a 240 Delay FIR Comb Filter.

misplacement can be investigated, in order to establish minimum performance criteria. The filter action is of course to subtract the wave from the previous revolution from that of the present. In the limit case the number of samples in a wave is miscalculated by one sample. Therefore, the subtracted wave will have a phase shift equal to one sample period. This can be evaluated in terms of the number of samples per wave. If there are m samples per wave the phase shift in the limit case will be equal to $2\pi/m$ radians. The results of these calculations are shown in Figure 3-8. The magnification factor of the filter is plotted versus the number of waves per cycle for maximum notch misplacement.

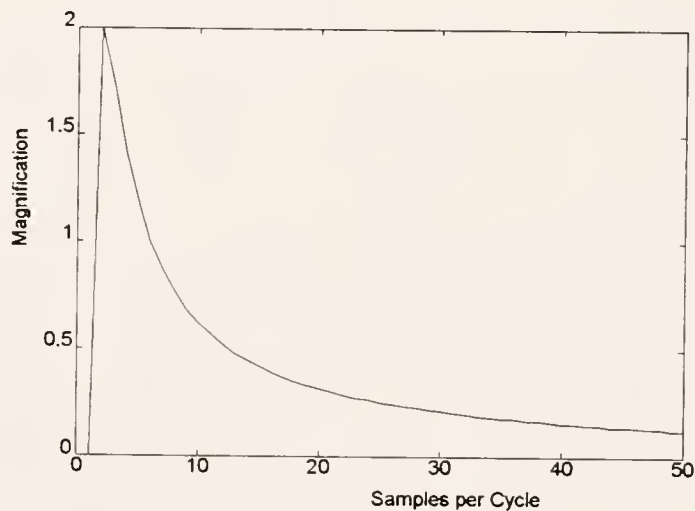


Figure 3-8 Magnification versus Number of Samples per Cycle at Maximum Notch Misplacement.

Several points of this plot are of interest. With only two samples per wave the phase shift is π or 180 degrees; when the two out of phase waves are subtracted the resulting output is two times the amplitude of the input wave. The magnification factor is equal to one when there are 6 samples per wave, or a phase shift of $\pi/3$.

Therefore any wave with fewer than six samples per cycle may be magnified by the revolution referenced filter. This is the lower limit performance of the filter, that is with the spindle speed such that there is a one sample miscalculation of the spindle speed. If the spindle is operating at a speed with less notch misplacement, ie. closer to an integer number of samples per revolution, the performance will improve with the limit being at an integer number of samples or synchronous sampling.

These results can be checked against the previously plotted signal for a spindle speed of 2490 RPM. For a sampling frequency of 10,000 Hz, the frequency of a wave with six samples per cycle is 1666.7 Hz. By comparing the plot of the unfiltered to the filtered signal it can be seen that the content above this frequency, ie. fewer than six samples per wave, is in fact amplified by the filter.

This can also be used to set the minimum required sampling frequency for a set filtering action. For example, if the frequency content of a signal must be attenuated by a factor of two (magnification factor equal to 0.5) through a frequency of 5000 hz., there must be 13 samples or more per wave. Therefore to have 13 samples per wave at 5000 Hz, the sampling frequency must be at least 65,000 Hz.

The Effects of Speed Variation

The effects of spindle speed variation on the filtering action are demonstrated through the use of a time domain cutting simulation that allows nonconstant spindle speed. The speed variation is used to simulate spindle speed sag upon entering a cut. The simulation allows the user to input the average spindle speed, a percent speed

variation and the number of revolutions of the cutter to be simulated. It also allows input of other cut information, such as the number of teeth on the cutter, radial and axial immersion. The speed variation is then set equal to a full cycle of a cosine wave, with length equal to the entire simulation time, and amplitude equal to the percent speed variation times the average speed. The overall speed is then the sum of the average speed and the speed variation. The simulation also generates a series of the integer number of samples per revolution for use in the revolution referenced filtering.

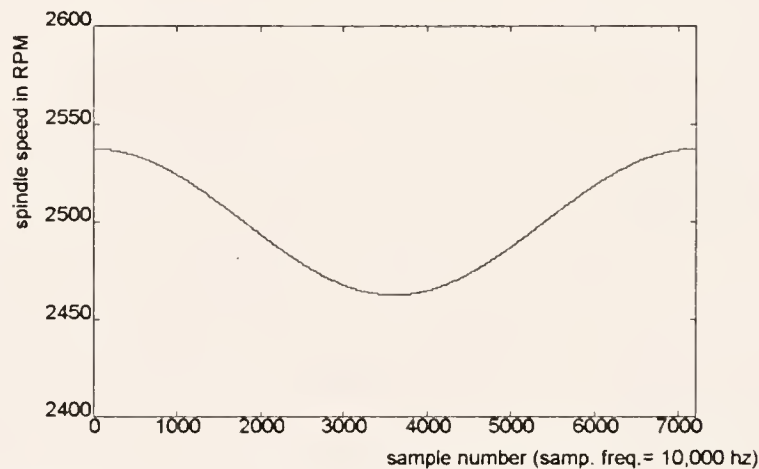


Figure 3-9 Speed Profile for 1.5% Speed Variation Over 30 Revolutions, Sample Frequency 10,000 Hz.

This speed profile is shown in Figure 3-9. The average speed is set to 2500 RPM and a speed variation of 1.5% is applied, and the simulation is run for 30 revolutions of the cutter with a sampling frequency of 10,000 Hz. This gives an average speed variation of 0.1% per revolution. Similar speed variations have been observed for experimental data (specifically those gathered at Giddings and Lewis).

Three simulations are used to first demonstrate the effect of the speed variation alone. The first is a steady state cut with a spindle speed of 2500 RPM, the second is a simulation with a 1.5% speed variation as shown in the previous figure, and the third has a large speed variation of 7.5 % or 0.5% per revolution. The effect of the speed variation on the spectral content is shown by plotting the magnitude of the Fourier transform of the signals. The window of the transform is 1024 points long, therefore encompassing approximately 4.25 revolutions of the cutter. The data window transformed is selected between points 1001 and 2024 of each simulation. It can be seen in the previous figure that this data is located during the period of maximum speed variation for the simulation. Each of the simulations is run with four cutting teeth and a radial immersion of five percent. The very low radial immersion is used to assure that there will be frequency content throughout the spectrum.

Figures 3-10A, B and C show the transformed data for the steady state, 1.5% speed variation and 7.5% speed variation, respectively. In the steady state milling condition the tooth frequency harmonics are clear and sharp, but as the speed variation increases, there is a blurring of the spectral peaks. This is particularly apparent in the 7.5% speed variation case, in which the upper harmonics become so blurred as to be lost and the lower harmonics widen their peaks as compared to the steady state case. The action of the revolution referenced filter is shown in Figure 3-11 A and B. This figure shows the spectrum of the filtered signal for the 1.5 and 7.5% speed variations. The figures are plotted to the same scale as the unfiltered signals for ease of comparison. The filtered spectrum for the steady state case is the same as shown in

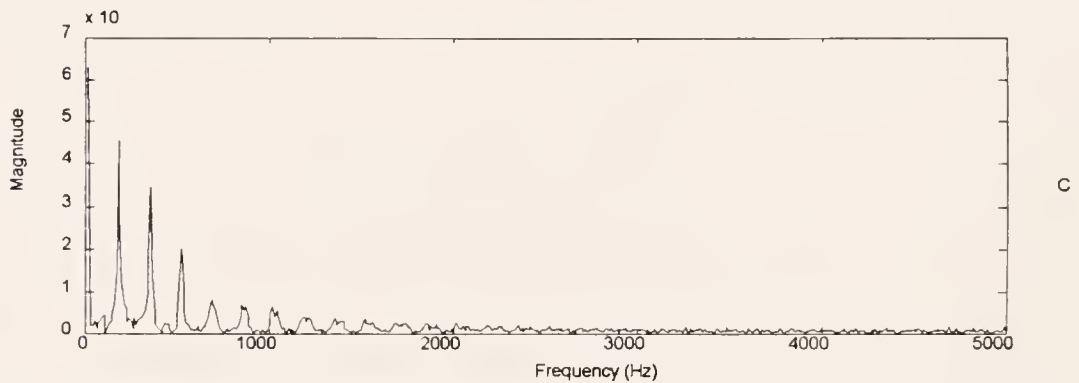
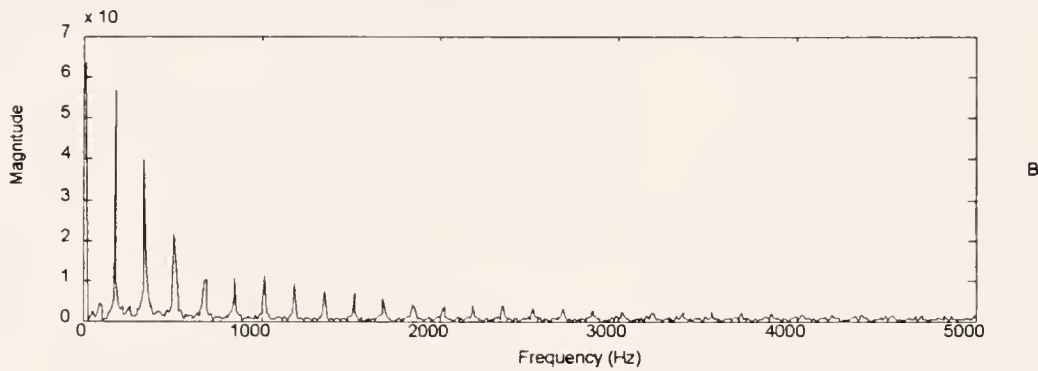
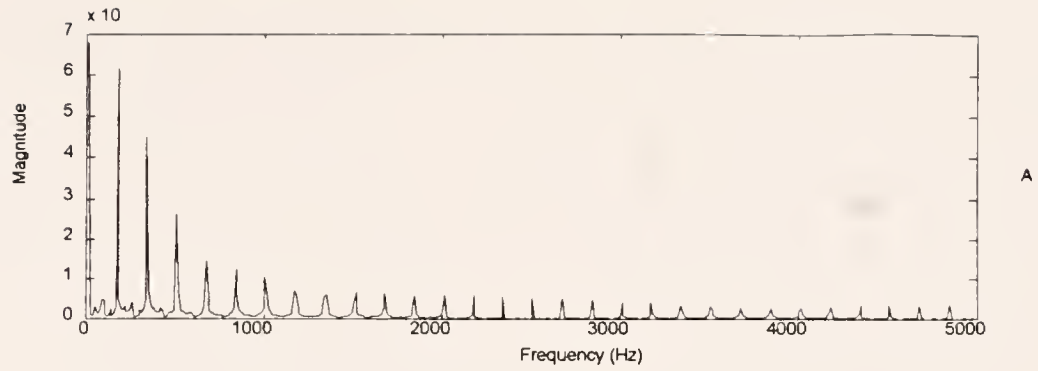


Figure 3-10 Spectrum of Simulated Cutting Force Signals for A) Steady State B) 1.5% Speed Variation C) 7.5% Speed Variation. Average Speed 2500 RPM, 4 Teeth, 5% Radial Immersion.

Figure 3-5. The filter uses the count signal generated by the simulation to update its delay value once per revolution. Therefore, the action of the filter on the simulated signal is identical to that of the actual revolution referenced filter.

The filter shows excellent performance on the 1.5% speed variation case, eliminating all noticable peaks. The 7.5% speed variation case still has some low frequency content, but it has been greatly attenuated. The performance of the filter in

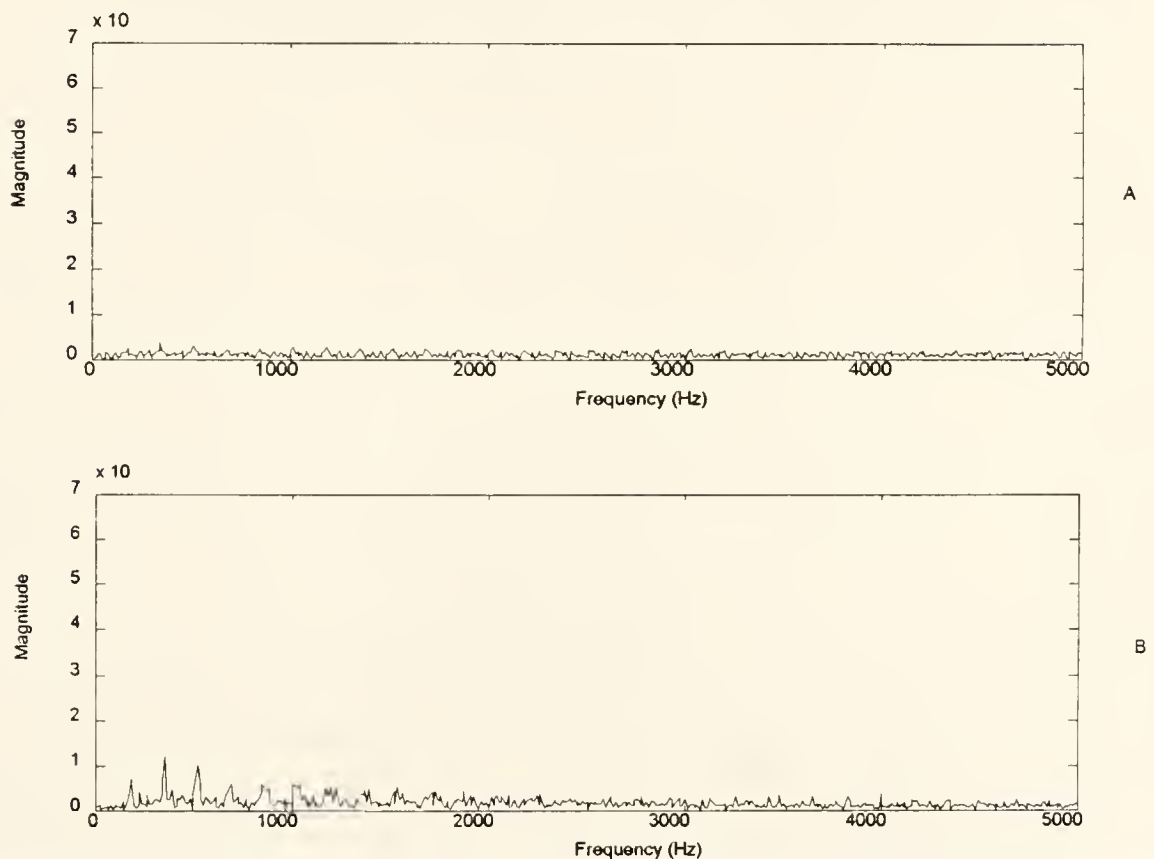


Figure 3-12 Spectrum of Simulated Cutting Force Signals Filtered Through the Revolution Referenced Comb Filter A) 1.5% Speed Variation B) 7.5% Speed Variation. Average Speed 2500 RPM, 4 Teeth, 5% Radial Immersion.

the higher frequency ranges should diminish first, as the phase of these signals would show the greatest variance with speed variation. However, as was seen in the

unfiltered spectra, the natural blurring of the high frequency content due to the averaging effect of the Fourier Transform effectively removes these signals.

Although simple to implement and very effective, the revolution reference comb filter has two main draw backs. First, because of the comb action with the run-out as the primary signal, all of the harmonics of the run-out are filtered from the sound signal. As previously mentioned in the introduction, these are not necessarily stable milling speeds. Therefore the filter can inadvertently filter a valid chatter signal. Recently it has been shown by Smith and Winfough [8] that for a four flute cutter, one of these notches falls such that chatter at the most unstable speed is filtered out. The second area of concern is the relatively wide filter notches. The wide notches filter what may be a valid chatter signal that is near a run-out harmonic. The revolution referenced comb attenuates one third of the frequency spectrum to half power or lower, when compared to the power of the maximum transmitted signal.

CHAPTER 4

FILTER MODIFICATIONS

Two modifications to the revolution referenced comb are presented here. First the addition of poles to the filter to make an IIR comb filter which narrows the filter notches and reduces the likelihood of filtering a valid chatter signal that is near a run-out harmonic. The second modification splits the signal into two signals and performs separate filtering functions on each signal portion, then recombines them. This is done to avoid filtering all of the run-out harmonics and thereby avoid filtering a valid chatter signal that may exist at one of the upper run-out harmonics, while still removing the lower harmonics to avoid false triggering of the system. The specific detail and the results of the application of the two modifications is presented in the following sections. Further the existence of the run-out harmonics in the spectrum is explained and consequently, the number of harmonics that should be filtered is determined.

Infinite Impulse Comb

The first modification to the revolution referenced comb is the addition of poles to the filter to make an IIR comb. This is done to narrow the filter notch width to avoid filtering of a chatter signal that is positioned in the frequency domain near one of the filter notches.

The revolution referenced (FIR) comb is modified by placing a series of poles at the same frequency locations as the zeros of the FIR comb but slightly inset from the unit circle. The transfer function of this filter can be written as follows:

$$H(z) = \frac{(1 - z^{-N})}{(1 - R z^{-N})}$$

where N is again the number of samples per revolution and R is the radial location of the poles. The radius of the poles must of course be less than one for stability. The difference equation used to implement this can be written:

$$Y(n) = X(n) - X(n-N) + R Y(n-N)$$

It is seen that the implementation requires only one more multiplication and one more addition per output sample than the FIR comb.

The effect of the added poles is shown in Figures 4-1 and 4-2. Figure 4-1 shows a pole-zero plot of the IIR comb filter with the poles inset along the radial lines

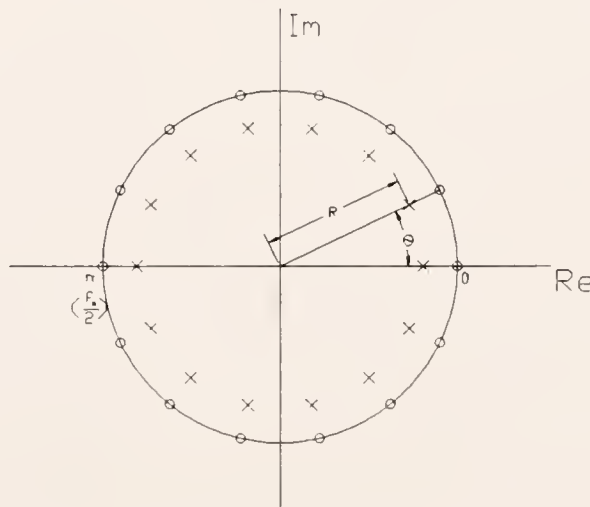


Figure 4-1 Pole-Zero Plot of 15th Order IIR Comb Filter.

through the zeros. The magnitude response transfer function is shown as Figure 4-2. Four curves are plotted, with radial locations of 0, 0.5, 0.9 and 0.95. The curve for radial location zero is the same transfer function as the FIR comb. As the radius is increased it can be seen that the filter notch widths can be narrowed or sharpened, thus attenuating less of the surrounding signal. Additionally the pass bands, between the zeros, are flattened, giving a much more even amplification of the transmitted signal.

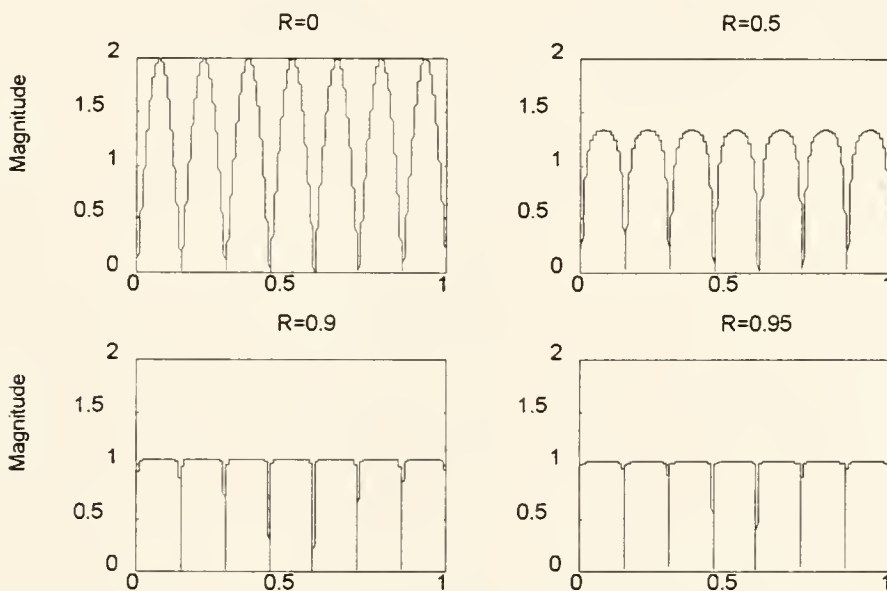


Figure 4-2 Magnitude Transfer Function of IIR Comb Filters with Various Radii Poles.

A concern of implementing the IIR comb with its narrow notches is the effect of spindle speed transients on the filtering action. The spindle speed may sag when entering a cut, and when this happens, the run-out frequency and its harmonics also sag. The FIR comb has been effective in these situations. The narrowed notches of

the IIR filter may allow the decreasing frequency to pass. If this proves to be the case in practice, the IIR filter can be modified to include a term to vary the radial location of the poles based on the amount of spindle speed variation. The radial location can be based on a simple linear formula as follows:

$$R = R_{\max} - m | (N - N_{\text{past}}) |$$

where N is the count number for the present revolution and N_{past} is the count for the previous revolution. The slope is the sensitivity to speed variation, and $R_{\max}$ sets the narrowest notch width, which is used during steady state cutting. By this method, when there was a speed transient the poles would be moved in towards the origin, thereby increasing the width of the filter notches. In the limit case the poles would be allowed to move to the origin (a limit would be needed to keep the radial location greater than or equal to zero) which would give the same transfer function and performance as the present FIR filter. As the speed stabilized, the poles would move out, narrowing the filter notches thereby attenuating less of the signal.

Although the filter notch widths will be reduced by the addition of the poles, two problems of the FIR comb are still present in the IIR comb. The filter still eliminates all the run-out harmonics, possibly filtering a valid chatter signal. Because the notches have been sharpened this is somewhat less likely with the IIR comb. There is still a misplacement of the zeros and poles due to the discrete delay required by the comb filter.

Results for the Infinite Impulse Comb Filter

The performance of the Infinite Impulse Response comb filter is first demonstrated on simulated stable milling cuts. The signal used in demonstrations of the revolution referenced comb is also used here for direct comparison. That signal is of a four tooth cutter, milling at 2500 RPM at 5% radial immersion (to assure frequency content throughout the spectrum) and the sampling frequency is 10,000 Hz.

The Fourier transform of the unfiltered signal, the signal filtered through the FIR comb, and the signal filtered through IIR combs with pole radii of 0.7 and 0.9, are shown in Figure 4-3 A, B, C and D, respectively. Note that for direct comparison the filtered signals are plotted on the same scale as the unfiltered signal. The filtered signal from the FIR comb and the IIR comb with pole radius of 0.7 are replotted on an expanded scale to better reveal their form in Figure 4-4.

It can be seen in the figures that the FIR comb with its wider notches more effectively removes the tooth frequency harmonic components from the signal. In the expanded plot the peaks are seen to be spread by the FIR filter, while for the IIR, they are sharper. The content in either filtered case is seen to be greatly reduced when compared to the unfiltered case.

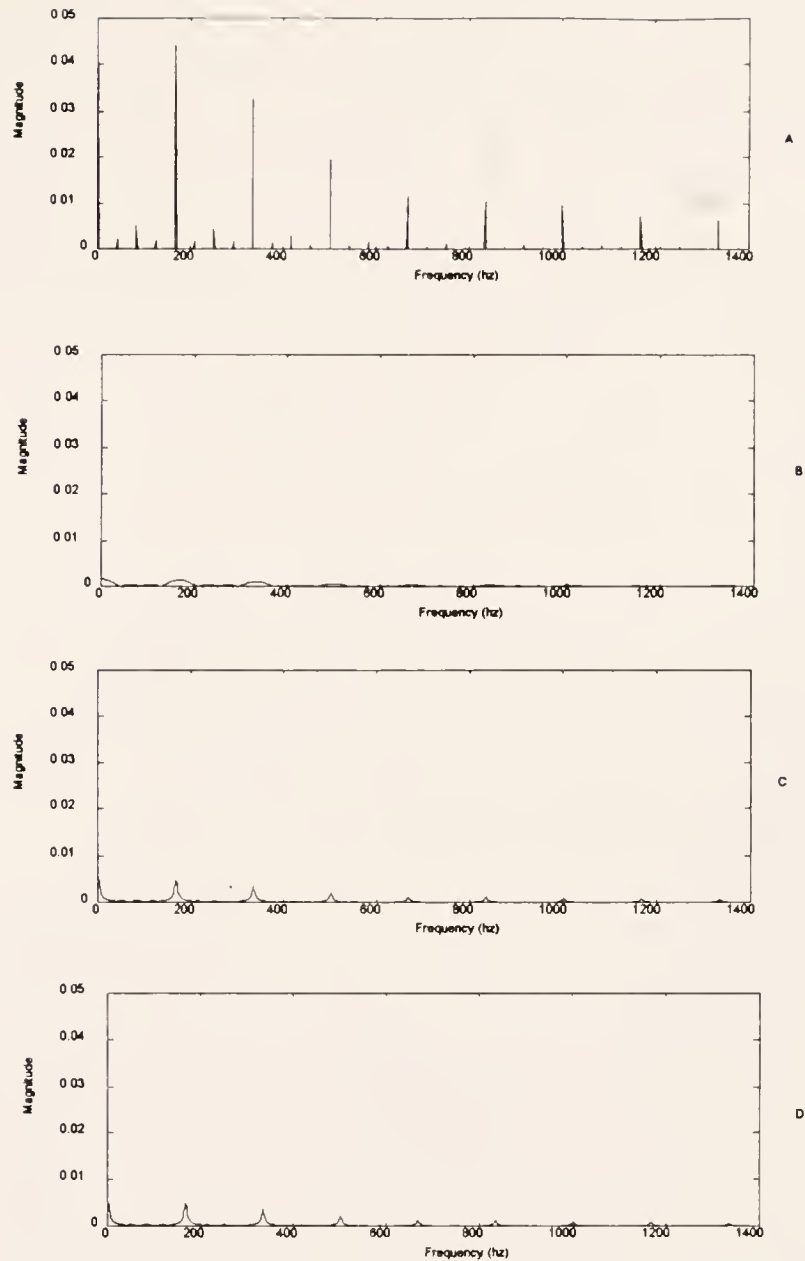


Figure 4-3 Fourier Transform of Simulated Cutting Force Signals 2500 RPM, 4 Flute, 5% Radial Immersion Sampling Frequency 10,000 Hz. A) Unfiltered Signal B) FIR Comb Filtered C) IIR Comb, Pole Radius = 0.7 D) IIR Comb Pole Radius = 0.9.

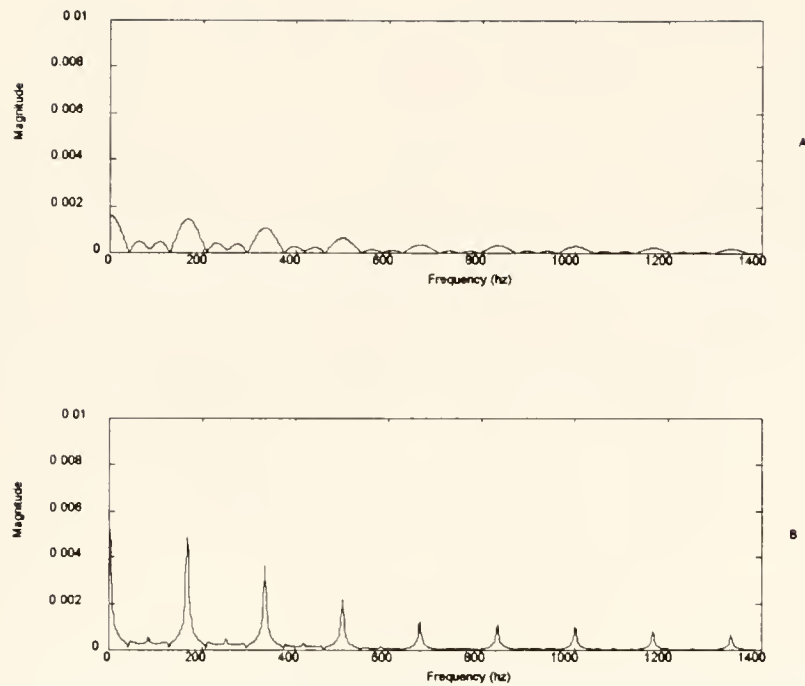


Figure 4-4 Expanded Frequency Plot of Filtered, Simulated Force Signals A) FIR Comb Filter B) IIR Comb Filter, Pole Radius = 0.7.

The effect of the IIR comb on an unstable cut with the chatter frequency very close to a run-out harmonic is now compared to the performance of the FIR comb in similar conditions. The effects are demonstrated on a simulated unstable milling cut with a single degree of freedom. The natural frequency of the mode of the system was varied to enable placing the chatter near a run-out harmonic of the cutting force signal. The simulated cut is for an eight tooth cutter operating at 2500 RPM with 25 percent radial immersion. The sampling frequency for the signal is 10,000 hz. The natural frequency of the chatter mode was first chosen equal to 579 hz., resulting in chatter very near the 14th run-out harmonic which is located at 583.3 hz. The chatter frequency was 582 Hz and the spectrum shown has a resolution of slightly over 1.3

Hz. The spectrum of the unfiltered signal is shown in Figure 4-5. The chatter is seen near the 14th run-out harmonic.

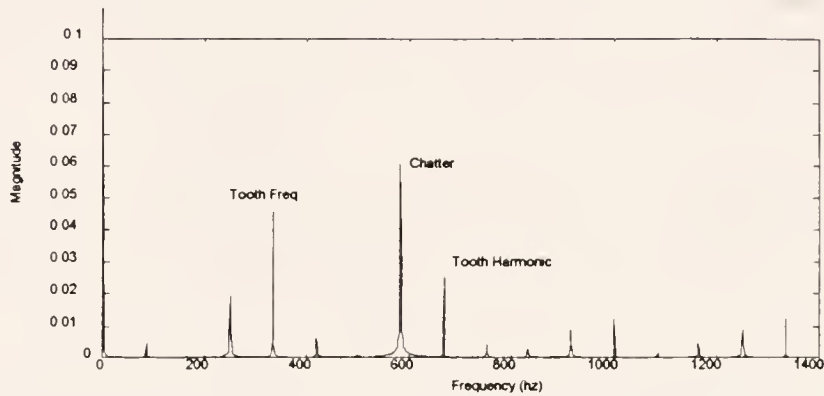


Figure 4-5 Spectrum of simulated Force Signal Unstable Milling, 2500 RPM, 8 Insert Cutter, 25% Radial Immersion, Natural Frequency 579 Hz.

The effect of both the FIR and IIR comb, with pole radius equal to 0.85, are shown in Figure 4-6 A and B, respectively. The FIR filter is seen, as expected, to

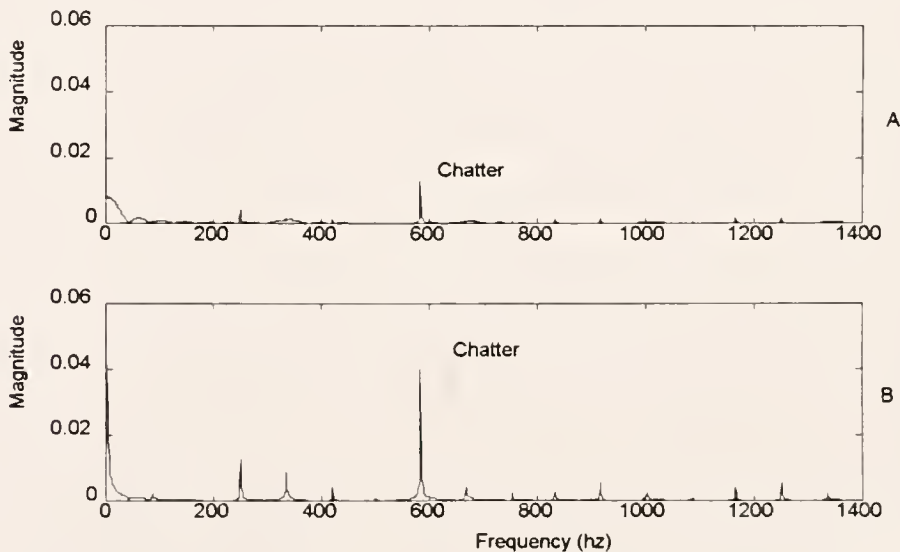


Figure 4-6 Cutting Force Spectra for Natural Frequency Equal to 579 Hz. Filtered by A) FIR Comb B) IIR Comb.

more thoroughly remove the tooth frequency components than the IIR filter. However, the FIR filter also suppresses the chatter signal more than the IIR filter: the IIR filter leaves more than twice the power in the chatter signal.

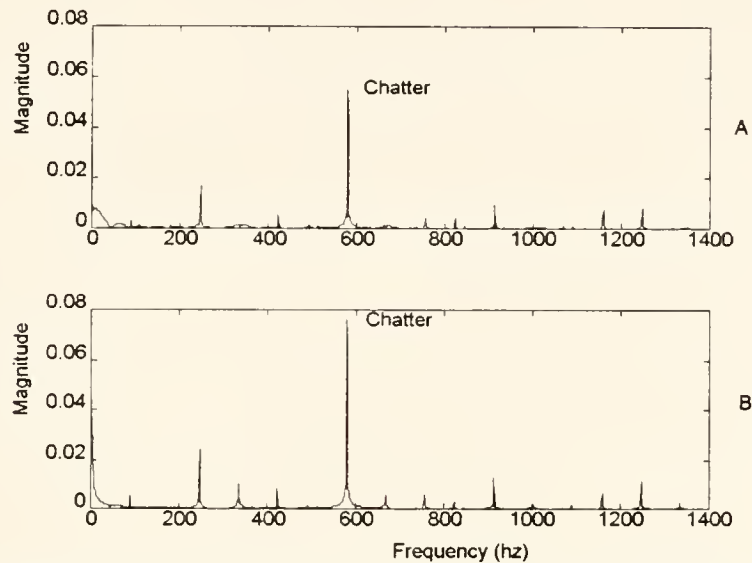


Figure 4-7 Chattering Force Spectra for Natural Frequency Equal to 575 Hz. Filtered by A) FIR Comb B) IIR Comb.

The performance of the filters is demonstrated as the chatter frequency moves further from the run-out harmonic. The natural frequency of the system is decreased to 575 hz. The spectra for the two filtered signals are shown in Figure 4-7. The FIR filter still slightly suppresses the chatter signal. The chatter signal in this case is now completely out of the IIR filter notch and is in fact slightly amplified when compared to the unfiltered signal. This slight improvement of the IIR filter as compared to the FIR is only seen in the very limited situation of the chatter being closely aligned with a run-out harmonic.

The Effect of Notch Misplacement

Because of the narrow notch width of the IIR comb as compared to that of the FIR comb, the effect of notch misplacement will be much more prevalent for the IIR filter. The effect of notch misplacement is investigated using the same simulated signal as used to demonstrate the effect on the FIR comb. The signal is of a simulated milling cutter with four teeth cutting at 2490 RPM, and a sampling frequency of 10,000 Hz. This gives just under 241 samples per revolution, such that the filter will be operated with a delay equal to 240 samples and have nearly maximum notch misplacement. In order to assure frequency content at the tooth harmonics throughout the spectrum, the cut is simulated to have radial immersion of 5%. Figure 4-8 shows the spectra of the simulated signal. Figure A is the unfiltered signal, B is the signal filtered through the FIR comb, and C and D are filtered through IIR comb with pole radii of 0.7 and 0.9 respectively.

The IIR combs are both seen to perform poorly under the condition of notch misplacement. The spectrum of the filtered signal from the comb with pole radius of 0.9 is almost identical to the unfiltered signal. This is due to the notch misplacement being greater than the notch width of the filter. Hence, the filter notches completely miss the run-out harmonics they are meant to filter, leaving the signal essentially unfiltered.

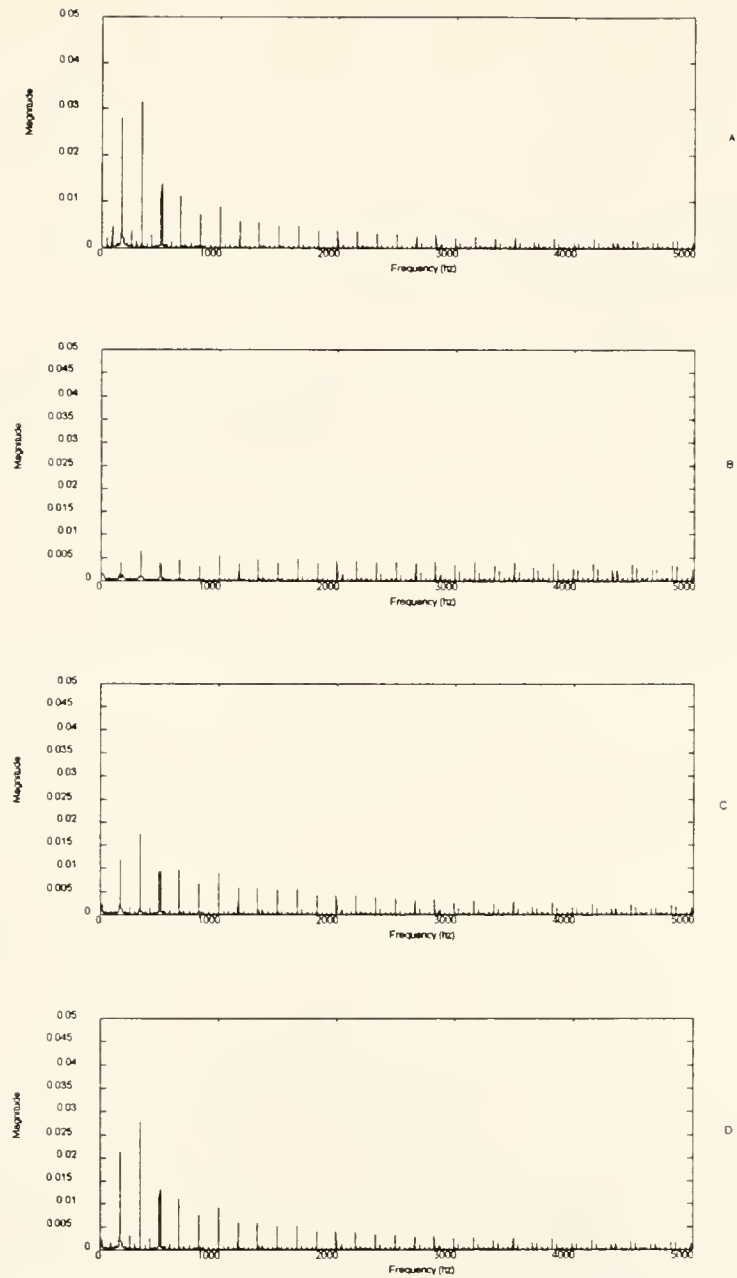


Figure 4-8 Effects of Notch Misplacement A) Unfiltered Signal B) FIR Comb C) IIR Comb, Pole Radius = 0.7 D) IIR Comb, Pole Radius = 0.9.

Conclusions for IIR Comb

Due to its narrow notch width, the IIR comb filter was shown to give an advantage when filtering a signal with the chatter component very close to a run-out harmonic. The narrow notch width, however, introduces a serious disadvantage to the IIR filter as compared to the FIR comb. Due to the narrow notch width, the effect of notch misplacement, from using a non-integer number of samples in a revolution, is much more severe. The notch misplacement can cause the IIR filter to be completely ineffective in filtering the tooth and run-out harmonics in the present system, thus it is not recommended for use.

Steps could be taken to reduce or eliminate the problem of notch misplacement, thereby allowing the use of the IIR filter and gaining its advantages. To reduce the notch misplacement in the present system, the sampling frequency may be increased. This increase in sampling frequency means there are more samples per spindle revolution; therefore, miscounting the revolution by one sample causes a smaller notch misplacement. The notch misplacement is directly proportional to the duration of the sampling. Thus doubling the sampling frequency will reduce the notch misplacement by a factor of two. This would allow the use of a IIR filter with notch width equal to one half of the FIR notch width while still attaining the same performance for tooth and run-out harmonic removal. The narrower notches would filter less content around each harmonic, giving improved performance when the chatter was closely located by one of the filter frequencies.

The problem of notch misplacement may be completely eliminated by changing the sampling system. The system may be changed from a time based sampling system to a revolution synchronized sampling system. By synchronizing the sampling with the spindle rotation, there is always an integer number of samples in the revolution (a method to do this is explained under other filtering methods at the end of this section). With an integer number of samples there is no notch misplacement and the full advantage of the narrow notch width of the IIR filter can be realized.

High Pass Low Pass with Combs

In order to avoid the filtering of all the run-out harmonics and possibly filtering a valid chatter signal, a two path filtering scheme is proposed. The author would like to credit Russ Walters for first suggesting this filtering method. By first passing the sampled signal through both a high pass and a low pass filter, the signal can effectively be split into two signals. One carrying the low frequency range, and the other, the high frequency range. The low frequency signal can then be filtered with the presently used FIR comb or the proposed IIR comb to remove the run-out and its harmonics. The high frequency signal can be filtered through a similar comb with the delay shortened by dividing the original delay by the number of teeth on the cutter (this new delay must be rounded to the nearest integer). The new comb removes the tooth frequency and its harmonics from the high frequency signal. The two signals may now be recombined through addition. In this way, the run-out and its harmonics

are removed up to the frequency of the cut off of the low and high pass filters, and the tooth frequency harmonics are removed throughout the entire frequency range.

A signal flow diagram of this process is shown in Figure 4-9. The upper path is through the low pass filter and revolution referenced comb, the lower path is through the high pass filter and tooth harmonic filter. The figure shows the transfer function of each path up to the point at which it is shown. The overall transfer function is reproduced as Figure 4-10. Several parameters can be varied to adjust its exact shape.

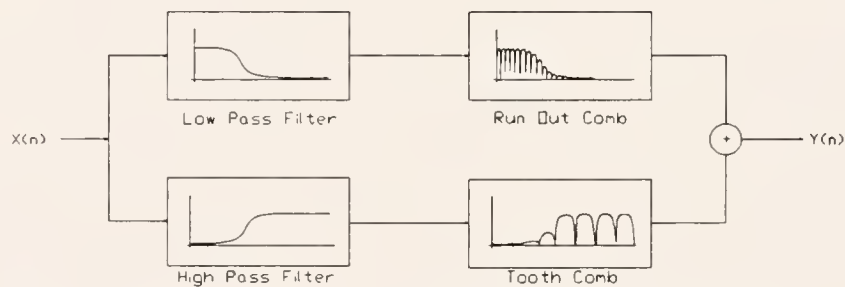


Figure 4-9 Flow Diagram for High-Low Pass Filtering with Combs.

First, the shape of the transition region between where all the run-out harmonics are removed and the area where only the tooth harmonics are removed, may be shaped by adjusting the filter parameters of the low and high pass filters. This may be limited by the relative phase of the signals. The frequency location of this transition is positioned with the cutoff frequency of the filters. This cutoff may be positioned by two methods. It may be positioned at a set frequency, or it may be positioned to filter a set number of run-out harmonics. If the filters are positioned at a

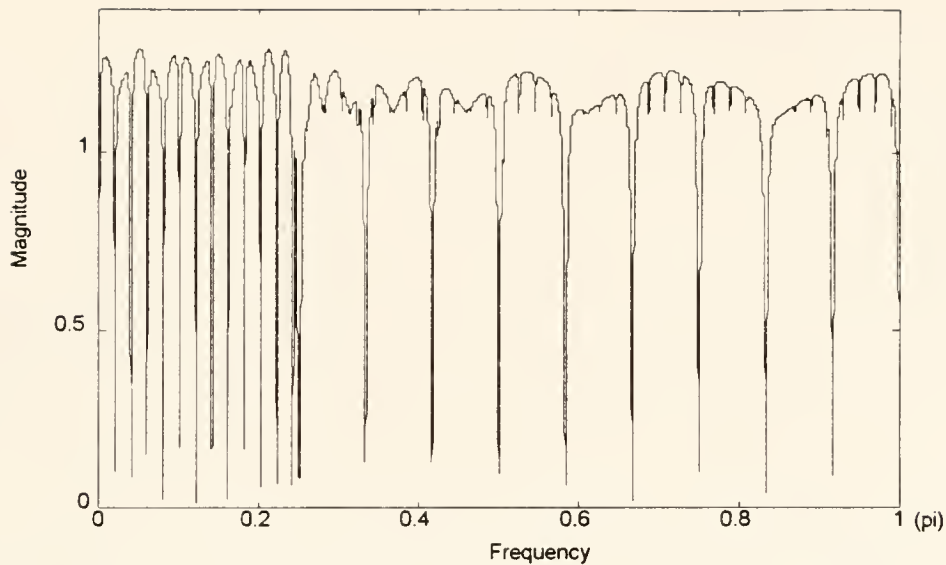


Figure 4-10 Magnitude Transfer Function for High-Low Pass Comb Filter System.

set frequency they are stationary with respect to the spindle speed. That is, the filter coefficients may be calculated at the start of the program and are not affected by the spindle speed. If the filter is set to remove a set number of run-out harmonics, the filter coefficients must be recalculated for a change in spindle speed. It is not proposed to vary these with a sag in speed, only during a commanded spindle speed change. By varying the order of the filters the width of the transition between the two filter regions may be adjusted. A low order filter will have a broader transition region.

The sharpness of the notches in each comb filter can be varied independently. Therefore, the notches applied to the tooth harmonics can have their poles at a lower radial location than those of the run-out filter. Because these harmonics are higher in frequency region, the misplacement of the zeros is more severe. Additionally, these positions are not possible chatter frequencies, so there is less danger of removing valid chatter signals in the area.

Two main concerns must be addressed in the implementation of the high-low comb filtering method. The first concern is that of the effect of rounding the delay number in setting the tooth harmonic comb. To set the delay used in the tooth harmonic filter, the delay used in the run-out filter must be divided by the number of teeth on the cutter. The number of samples collected in one revolution (the delay number used in the run-out filter), will most probably not be evenly divisible by the number of teeth on the cutter; this number must be rounded to be used in the new comb. The effect of this rounding on the location of the zeros must be investigated. The second concern is which run-out harmonics to remove from the signal.

The issue of rounding the count number will be addressed first. Rounding of the delay number produces a notch misplacement in addition to the misplacement from using an integer delay number to approximate the revolution length. This misplacement can be expressed as a phase shift between the two waves, that is the present wave and the one from the previous tooth period that is being subtracted to perform the filtering. As previously addressed, the phase shift due to miscounting the revolution length is at a maximum with a shift of one sample period. Therefore, the misplacement of the primary tooth frequency notch is,

$$phase\ shift = \frac{m (2\pi)}{N}$$

where m is the number of teeth on the cutter and N is the number of samples per revolution. For a particular notch, this is the same phase shift and therefore notch misplacement as would be achieved with the run-out comb, ie. the forth notch of a

run-out comb would have the same misplacement as the primary notch for a tooth frequency comb of a four tooth cutter.

When applying the tooth frequency comb an additional phase shift is introduced due to the rounding. The tooth frequency comb is essentially a first difference method, therefore the delay used must be equal to the tooth period. This delay is obtained by dividing the number of samples per revolution by the number of teeth on the cutter. In order to perform the filtering without the use of interpolation of the data points, this delay must be an integer. The rounding of the delay number to the nearest integer value may produce, at the limit, a phase shift of one-half of a sample period. Therefore the total phase shift of the waves due to the miscounting of the revolution length and the rounding error can be expressed:

$$phase\ shift = \frac{m (2\pi)}{N} \pm \frac{m \pi}{N}$$

where the first term is due to the revolution miscount and the second term is due to the rounding error. It can be seen that the rounding may improve the placement or further misplace the filter notch, depending on the direction of the rounding.

However, because the misplacement direction due to the miscounting of the revolution length is not known, for the limit condition it must be assumed that the rounding will further misplace the filter notches. Therefore the limit notch misplacement with rounding will be

$$phase\ shift = \frac{m (3 \pi)}{N}$$

This is the phase shift for maximum miscounting of the revolution, miscount by one sample, and maximum rounding error. This is a fifty percent increase in the misplacement of the filter notches when compared to the corresponding notches of a run-out comb filter. To achieve equal performance for the tooth frequency comb to that of the run-out comb, the sampling frequency would have to be increased by fifty percent. By this method Figure 3-8 may be used to establish system requirements for a specified performance of the tooth frequency comb just as was done for the run-out comb.

The second subject of concern when applying the high-low pass comb system is to determine which run-out harmonics are to be filtered from the spectrum. As mentioned earlier, the run-out harmonics (not including the tooth harmonics), are possible chatter frequencies. Therefore, only those harmonics which may cause false triggering during stable cutting conditions should be filtered. To make this determination, it is first required to understand the source of the run-out harmonics and their expected distribution in the spectrum.

The first thing to note is the term "run-out harmonics" is a bit misleading. A tool run-out, when defined as an eccentricity of the tool, produces a sinusoidal variation of the chip load at a once per revolution frequency. This variation will result in content in the spectrum at the run-out frequency, and at frequencies separated from the tooth frequency and its harmonics by the run-out frequency, essentially side lobes

to the tooth frequency and its harmonics. The run-out does not produce a full population of run-out harmonics in the spectrum.

The effects of pure run-out are demonstrated in Figures 4-11 and 4-12. Figure 4-11 shows the time domain vector sum force pattern for an eight tooth cutter with a run-out of 50 % of the chip load. The cut is simulated for 20 % radial immersion. The figure shows the vector sum force pattern for four revolutions of the cutter. Figure 4-12 shows the spectrum of this force signal with the run-out component and the side lobing effect.

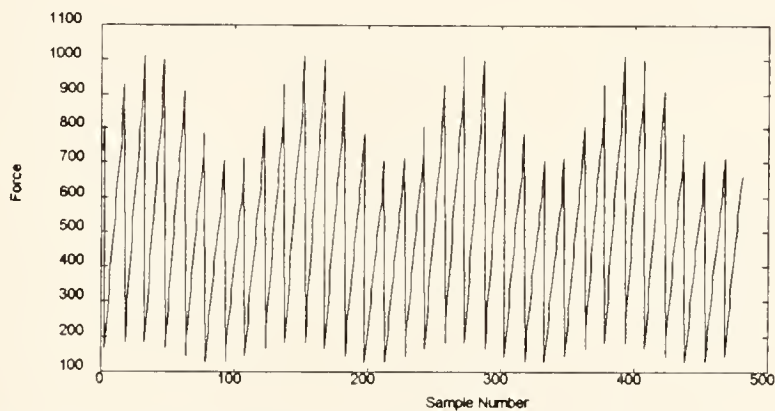


Figure 4-11 Force Pattern for Four Revolutions of an Eight Tooth Milling Cutter with 50% Run-out, Radial Immersion = 20%.

The existence of a full population of run-out harmonics is in fact due to tooth to tooth variation or tooth throw. Figures 4-13 and 4-14 show a vector sum force signal and its spectrum, respectively, for an eight tooth cutter with 50% tooth throw. This is defined as random radial variation of the tooth position with the maximum difference between any two teeth equal to 50 % of the chip load, and 20% radial immersion.

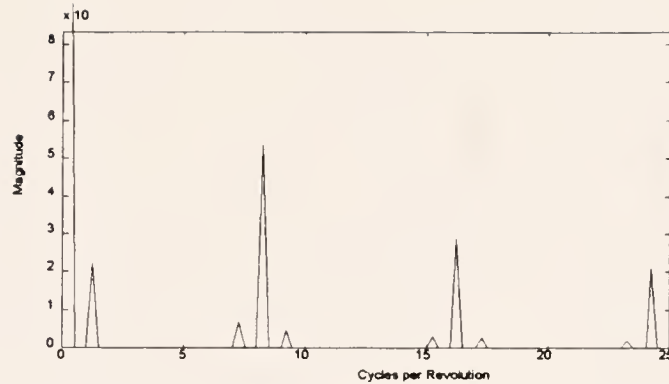


Figure 4-12 Spectrum of Force Signal from an Eight Tooth Cutter with 50% Run-out, Radial Immersion = 20%, Magnitude Plotted versus Cycles Per Revolution.

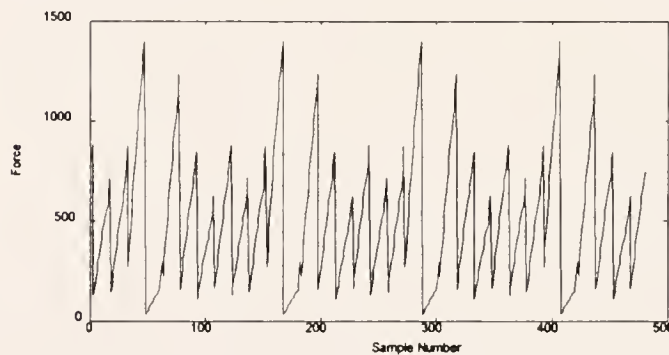


Figure 4-13 Force Pattern for Four Revolutions of an Eight Tooth Milling Cutter with 50% Tooth Throw, Radial Immersion = 20%.

This distribution of the run-out harmonics is controlled by the radial immersion of the cut and distinctive force pattern introduced by tooth throw. To illustrate this it is convenient to first look at the form of the force on the cutter. A diagram of a milling cutter with the tooth forces is shown in Figure 4-15. The tooth force is decomposed into its tangential, F_T , and radial, F_R , force components.

The tangential tooth force is assumed proportional to the area of the chip. If circular motion is assumed for the cutting tooth this tangential force can be expressed

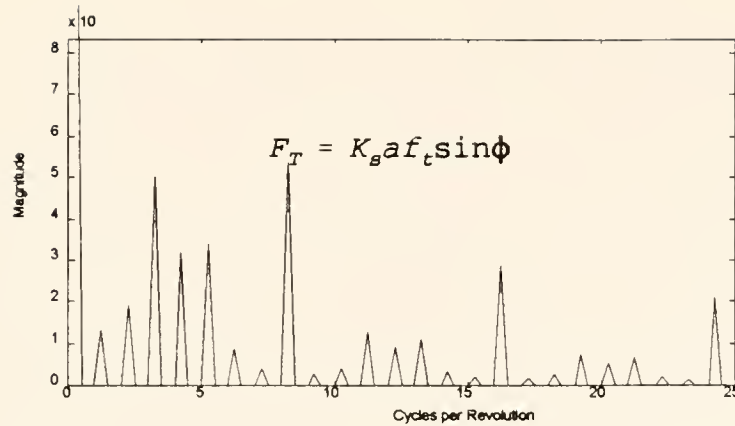


Figure 4-14 Spectrum of Force Signal from Eight Tooth Cutter with 50% Tooth Throw, Radial Immersion = 20%, Magnitude Plotted versus Cycles Per Revolution.

as,

where K_s is the cutting stiffness of the material, a is the axial immersion, and f_t is the feed per tooth. The radial force is assumed to be proportional to the tangential force and can be expressed as follows:

$$F_R = 0.3 F_T$$

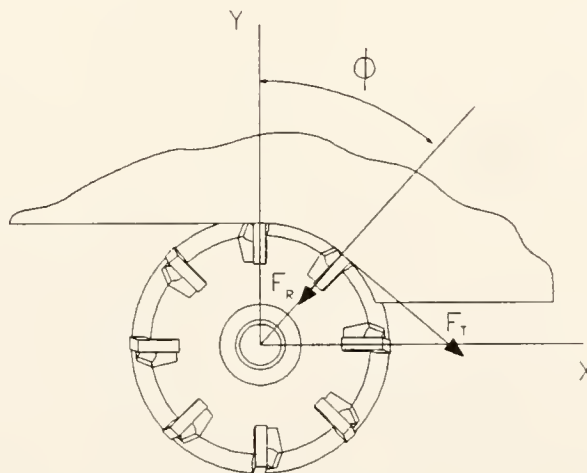


Figure 4-15 Tooth Forces on the Milling Cutter.

These tooth forces can be projected into the X and Y coordinate system to give the component forces.

$$F_x = F_T \cos\phi - F_R \sin\phi$$

$$F_y = F_T \sin\phi + F_R \cos\phi$$

The magnitude of the vector sum of the forces can be expressed as the square root of the sum of the squares of the force components.

$$F_{vs} = K_s a f_t \sqrt{1.09} |\sin\phi|$$

This force signal is a scaled and rectified sine wave. The scaling comes from the magnitude of the radial force relative to the tangential force. Because the cutter is only engaged from a possible position of 0 to 180 degrees, the vector sum force can be represented by a windowed sine wave as presented by Walters[9]. It should be noted that the vector sum and the tangential force have the same form and hence will have similar spectral properties. The force due to a single tooth is presented as a sine wave, scaled by a factor to account for the radial force component, the cutting stiffness, axial immersion and feed per tooth.

$$F_{vs} = c K_s a f_t \sin\phi$$

This wave is then 'masked' by a function that is equal to one during tooth engagement and equal to zero when the tooth is not engaged. This 'masking operator' is simply a windowing of the sine wave with the window length set equal to the duration of the

tooth engagement. This operation is shown in Figure 4-16, the sine wave and the windowing function are shown, then the product of the two is plotted for 25 % radial immersion.

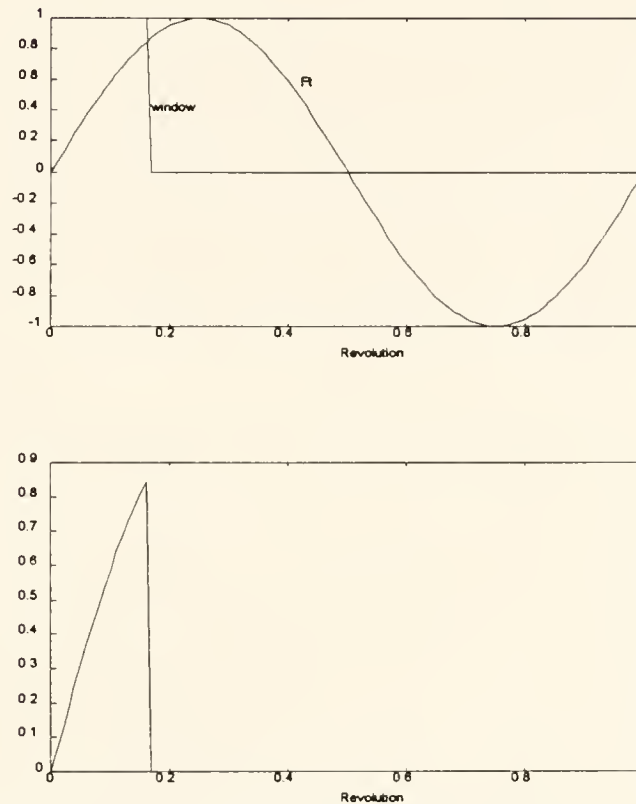


Figure 4-16 Single Tooth Force and Windowing Function for 25% Radial Immersion.

This windowing in the time domain is a multiplication operation equivalent to convolution in the frequency domain. Therefore, the frequency content of the single tooth force pattern is equal to the frequency content of the sine wave convolved with the content of the windowing function. The frequency content of the sine wave is of course a single line at the run-out frequency, and its mirror image at the negative of the run-out frequency. The frequency content of the square window is the sinc function as shown in Figure 4-17. The zeros of this function are located based on the

length of the window. The zeros are located at the frequencies of $1/T$, $2/T$, $3/T$... where T is the time duration of the square window.

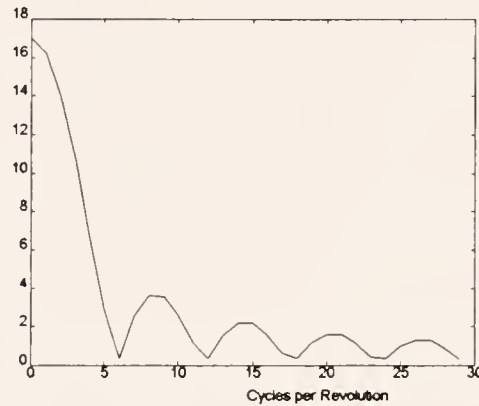


Figure 4-17 Frequency Content of a Square Window.

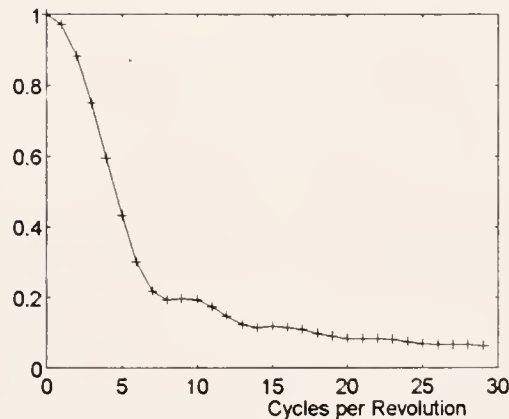


Figure 4-18 Frequency Content of a Single Tooth Force.

The convolution of the two signal portions is shown in Figure 4-18, the actual values are marked with crosses and the envelope of the maximum values is also plotted. This convolution results in a series of run-out harmonic lines with peak values at the marked positions. Because this discussion pertains to the distribution of

the run-out harmonics in the spectrum, the envelope of the harmonic peaks will be used.

Walters showed that for an ideal cutter with identical teeth, these frequency patterns for the individual tooth could be combined through the use of the 'force composition filter' to produce the overall cutting force signal. When all the teeth are equal, this filter produces zeros at all multiples of the run-out frequency and poles at the tooth frequency and its harmonics. In this way the force composition filter works as a comb filter, leaving only the tooth frequency and its harmonics in the composed signal. However, with tooth throw the teeth are no longer identical and there is another component added into the final signal. This added component is the source of the run-out and its harmonics in the spectrum.

The pattern of tooth throw is not just a single high or low tooth. Because a high tooth will remove more material from the cut than normal, the following tooth will remove less than normal by this same amount. The inverse being true for a leading low tooth. Therefore the pattern introduced by tooth throw is an increase and then a corresponding decrease of the force on successive teeth (assuming a high lead tooth). This pattern is superimposed on the ideal force pattern for equal teeth. Further, there is a throw between each successive tooth pair to make up the random variation of the teeth within the limits of the throw. Each of these throws is superimposed on the ideal cutting pattern. Superposition in the time domain is also

superposition in the frequency domain, hence the effect of the throw can be presented separately and then imposed on the force of the ideal cutter to generate the complete cutting force pattern.

The high low type force pattern of tooth throw is shown in Figure 4-19 for a 25% radial immersion cut with a four tooth cutter. The frequency content of this signal is shown in Figure 4-20, with the peaks of the harmonics marked.

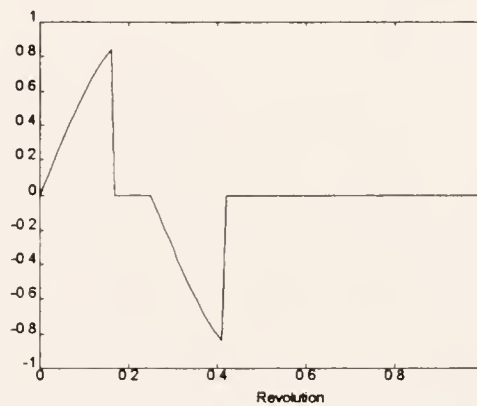


Figure 4-19 Tooth Throw Force Pattern for Leading High Tooth 25% Radial Immersion Four Tooth Cutter.

The frequency content is seen to have zeros at the tooth frequency and its harmonics, and the run-out harmonics spread between these zeros in a rounded fashion. This is indeed the pattern that is seen from simulation of cutting with tooth throw (Figure 4-14 at the start of this section) or due to cutter breakage which is equivalent to a tooth throw equal to the chip load. The position of the zeros is determined by the tooth spacing, ie. the number of teeth on the cutter and always produces zeros at the tooth frequency and its harmonics.

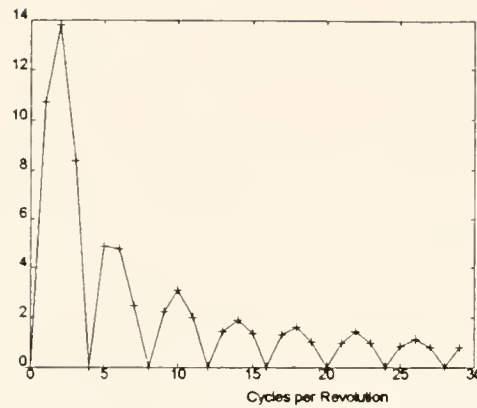


Figure 4-20 Frequency Content of Tooth Throw Force Pattern for 25% Radial Immersion Four Tooth Cutter.

Alternatively, the formation of the high low pattern from the single tooth force pattern can be thought of as passing the single tooth force through a filter which delays the signal by the tooth period and then subtracts it from the original signal. This is the action of a FIR comb filter with delay equal to the tooth period. The transfer function of such a filter was previously shown to be,

$$|H(z)| = |2\sin(k\Theta)|$$

where Θ is the digital tooth frequency. The magnitude of the transfer function is shown in Figure 4-21. The zeros of this function are located at the tooth frequency and its harmonics, and the maximum magnification is equal to two. This spectrum can be convolved with that of the single tooth to produce the expected pattern of run-out harmonics.

In order to accomplish our filtering task, determining which run-out harmonics to filter from the spectrum, it is of more interest to know the maximum possible distribution of the harmonics. If the frequency content due to each component of the

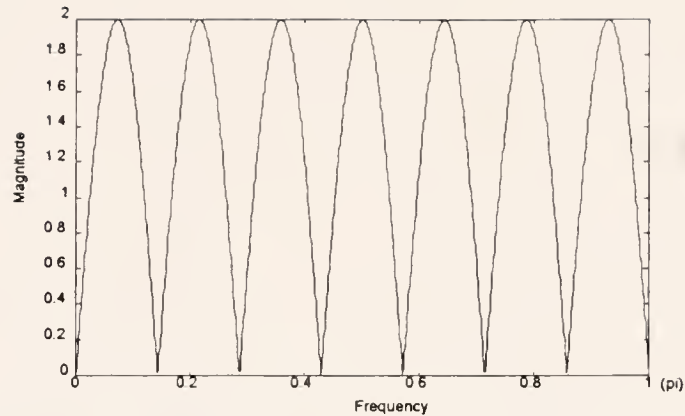


Figure 4-21 Magnitude Transfer Function of 15th Order Comb Filter.

tooth throw is combined, neglecting phase, an envelope of the maximum possible content is obtained. That is, the magnitude of the frequency of the positive tooth force is added to that of the negative. Further, the component due to the following tooth is simply the negative of a time shifted version of the positive force on the high tooth. Therefore the magnitude of the frequency content is the same for each component. The overall magnitude of the frequency content is thus two times the magnitude due to a single high tooth (the frequency content due to a single high tooth was shown as Figure 4-18).

This is demonstrated in Figure 4-22. The figure shows two times the magnitude of the frequency content for of a single high tooth for 25 % radial immersion. Also plotted is the content due to tooth throw of the same amplitude for a four and an eight tooth cutter at the same radial immersion. The line for two times the single tooth is seen to form a maximum envelope for the run-out harmonics, independent of the number of teeth on the cutter. The four and eight tooth cutter signals have zeros at their respective tooth frequencies, as expected.

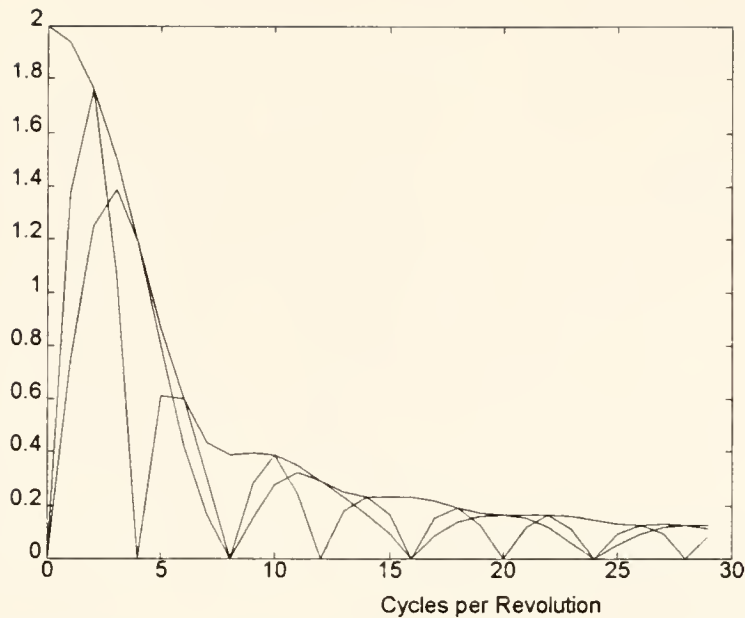


Figure 4-22 Frequency Content Two Times Single High Tooth, 25% Radial Immersion and Tooth Throw for Four and Eight Tooth Cutters.

The envelope of the distribution of the maximum possible run-out harmonic peaks can thus be defined by the radial immersion. This envelope is scaled by the actual amount of tooth throw but the form is determined only by the radial immersion. The envelope of the maximum possible run-out harmonics are plotted for 12, 25, 50 and 100 percent radial immersion as Figure 4-23.

It is obvious that the shallower the radial immersion the further the run-out harmonics will be spread into the spectrum. The point at which the run-out harmonic amplitude is $1/8$ of the maximum envelope height comes at approximately the 20th run-out harmonic for the 12% radial immersion case. This same point is located at the 13th harmonic for the 25% radial immersion case and the 8th and 3rd harmonics for the 50% and 100% radial immersion cases, respectively. Therefore, if the radial

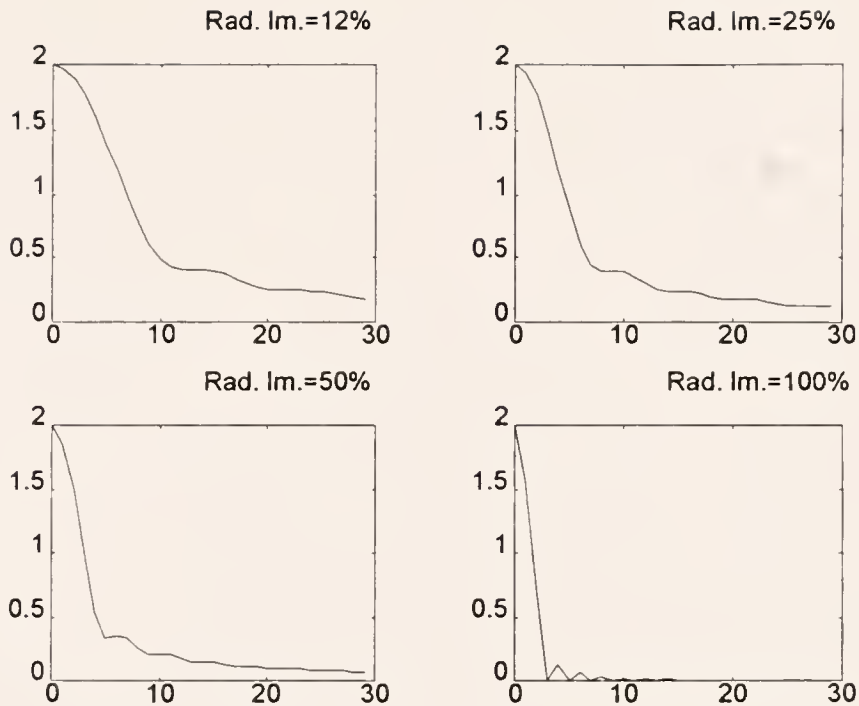


Figure 4-23 Maximum Possible Distribution of Run-out Harmonics in the Cutting Force Spectrum for Radial Immersions of 12, 25, 50 and 100%. Frequency in Cycles Per Revolution.

immersion of the cut is known, the transition point for the high low filter scheme can be chosen.

The radial immersion of the cut is generally not known. If a minimum radial immersion during the process is known, the transition point can be based on this value. The high and low pass filter coefficients are calculated based on the spindle speed in revolutions per second, multiplied by the harmonic number that is to be filtered. This gives the cut off frequency of the filters.

A further complication comes from the fact that this is the distribution of the harmonics in the force spectrum. The chatter detection system works on a sensed sound signal from the cutting process. This sound signal is highly dependent on the

displacement versus force transfer function. Therefore, harmonics of the run-out that lie near resonances of this transfer function are greatly amplified. This presents the most difficulties during very shallow radial immersion cuts, where the run-out harmonics are spread almost completely throughout the spectrum. Very high run-out harmonics past the high low filter transition may be amplified and cause false triggering of the detection system. The only solution in such cases is to filter all of the run-out harmonics, as is presently the practice.

Conclusions for High Low Pass Comb Filter

The form of the high low pass comb filtering system is by nature more applicable to cases with a large number of cutting teeth, such as face milling. In low speed milling the tooth frequency is substantially lower than the expected chatter frequency, ie. for stability the cutter is being operated in the higher number lobes. The run-out harmonic number is equal to the number of teeth times the tooth harmonic number. Therefore, with the cutter being operated such that the chatter can be expected around the upper tooth harmonics, ie. the third or forth, the power in the run-out harmonics will be small in this area. The cut off frequency for the high low pass filter may be set to filter the run-out harmonics up to the second tooth frequency harmonic, ie. for an eight tooth cutter the first 16 run-out harmonics. The run-out filter would then not be active in the region of the chatter, eliminating the possibility of filtering out the valid chatter signal.

The two path system is not applicable to the case of high speed milling with a low tooth count. In this case the cutter is being operated with the tooth frequency near the chatter frequency, in the region of the first stability lobe. Because the chatter is expected near the tooth frequency and there is a small number of teeth, the chatter will be in the area of the low number run-out harmonics. Therefore it would be impossible to locate a useful cut off frequency for the high low system below the area of the expected chatter. In such cases, and as is presently the practice, all the run-out harmonics must be filtered. The possibility of filtering the chatter may be reduced by narrowing the notches of the filter by going to the IIR comb. However, this would require that the issue of notch misplacement be corrected by going to a synchronous sampling system.

Notch Filters

As mentioned in the introduction, there are commonly other signals contained in the collected sound signal. These signals are due to noise sources such as electrical transformers, bearing noise or some other drive system noise. The signals may be stationary in frequency or related in some way to the spindle speed. Generally there are only a few noise sources and the signals are removed by placing individual notch filters at their respective frequencies. A second order notch filter is formed by a complex conjugate zero pair and a similar pair of poles located at the desired digital frequency; the pole is just radially inside the zero in the complex Z plane. A zero-

pole plot of a second order notch filter is shown as Figure 4-24. The rooted Z transfer function can be written:

$$H(z) = \frac{(z - 1e^{j\Theta}) (z - 1e^{-j\Theta})}{(z - R e^{j\Theta}) (z - R e^{-j\Theta})}$$

where R is the radial location of the pole. Theta is the desired digital frequency and the zeros are located on the unit circle.

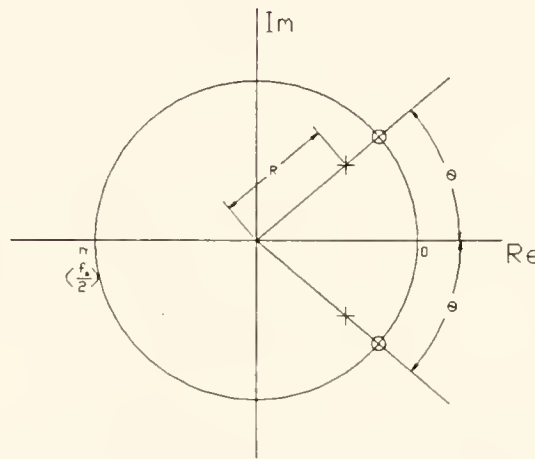


Figure 4-24 Pole-Zero Plot for a Second Order IIR Notch Filter.

The transfer function can be written in terms of real coefficients as follows.

$$H(z) = \frac{1 - 2 \cos\Theta z^{-1} + z^{-2}}{1 - 2R \cos\Theta z^{-1} + R^2 z^{-2}}$$

The digital frequency, Θ , is calculated based on the type of signal to be filtered. If the signal is not dependent on the spindle speed, Θ can be calculated from the definition of the digital frequency, namely

$$\Theta = 2\pi \frac{f}{f_s}$$

If the signal is based on the spindle speed, Θ can be calculated as a ratio of the run-out frequency that is used in the run-out comb

$$\Theta = a \frac{2\pi}{N}$$

where a is the desired ratio. The radius of the pole is set to adjust the band width of the filter. The radius is commonly around 0.9 or greater. A figure showing the effect of varying radial pole location is shown as Figure 4-25. A very sharp filter notch can be used, as the frequency of the noise can normally be accurately determined.

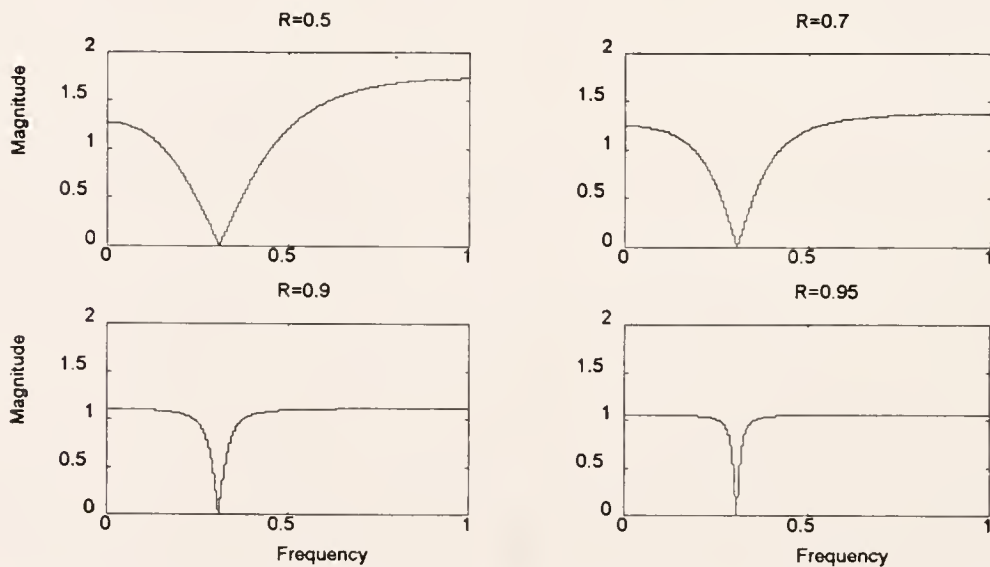


Figure 4-25 Magnitude Transfer Functions of Second Order IIR Notch Filters with Pole Radii of 0.5, 0.7, 0.9 and 0.95 Respectively.

Other Possible Filtering Methods

Three other possible filtering methods that show promise in this area will be discussed, although it is not proposed to fully investigate them in this work. These other techniques are described only as areas of future research. The three techniques are: 1) a series of cascaded notch filters, 2) multiple adaptive cancelers, and 3) using a phase lock loop and frequency synthesis to provide synchronous sampling.

Cascaded Notch Filters

The first technique presented is the use of a series of cascaded notch filters to remove the run-out, the tooth frequency and any number of desired harmonics of each. This system utilizes second order notch filters similar to those presently used to remove a few noise sources from the sound signal. A notch filter with the transfer function of,

$$H_1(z) = \frac{(z - 1e^{j\theta_1})(z - 1e^{-j\theta_1})}{(z - R_1e^{j\theta_1})(z - R_1e^{-j\theta_1})}$$

is formed for each signal component that is to be filtered. The signal is successively passed through each of the filters until all the components are removed. To pass the signal through multiple second order filters may seem inefficient, as opposed to convolving the coefficients of the filters together and then passing the signal through the single filter. However, digital signal processing chips, DSP's, are optimized to perform second order filtering operations. In fact, high order filters are often broken

down into a series of second order filters for implementation. To implement these filters their coefficients must be calculated, and must be done for each revolution in order to track spindle speed variation.

The transfer function in terms of real coefficients can be written.

$$H_1(z) = \frac{1 - 2 \cos \Theta_1 z^{-1} + z^{-2}}{1 - 2R_1 \cos \Theta_1 z^{-1} + R_1^2 z^{-2}}$$

or,

$$H_1(z) = \frac{b_{0_1} + b_{1_1} z^{-1} + b_{2_1} z^{-2}}{a_{0_1} + a_{1_1} z^{-1} + a_{2_1} z^{-2}}$$

It can be seen that three coefficients must be calculated for each filter, specifically b_1 , a_1 and a_2 . This also involves calculating a cosine for each set of coefficients. For the run-out and any harmonics (including the tooth frequency and its harmonics), the arguments of the cosines are conveniently related in that they are multiples of the run-out cosine argument. An efficient method for calculating the cosine values can then be formed from the trigonometric identity.

$$\cos(n\Theta) = 2 \cos(n-1)\Theta \cos(\Theta) - \cos(n-2)\Theta$$

The values can be calculated in a loop. Hence, only the cosine value of the run-out frequency need be calculated directly. An issue that would need to be investigated is the accumulated error in the coefficients due to calculating them in this loop fashion and its effect on filter performance.

The main advantage of using the cascade notch filters is that the zeros of the filter are located by the coefficients of the filter instead of simply by the delay as in the comb filter. By locating the filters with the coefficients, they may be continuously placed as opposed to the discrete placement of the delay located filter. Therefore, there is not the problem of the misplacement of the filter notches. Additionally, the specific signals to be filtered can be individually selected allowing any number of harmonics to be filtered and the filter for each to be selectively tuned for sharpness and overall attenuation. The disadvantage of the notch filter approach is that it is very computationally expensive. Considering a high speed milling application with a frequency range of interest of six thousand hertz, implementing the notch filters in the present DSP hardware would require approximately a 70 percent duty cycle. This is neglecting the calculation of the coefficients, the further transforming of the signal into the frequency domain and comparing peaks to a threshold to detect chatter. Therefore, it is not practical at this time to implement this filtering method. However, with the rapid advance of computing hardware, this may not be the case in the not too distant future. A final concern that would need to be investigated is the effect of changing the filter coefficients during the processing at a speed change. The effects of this time variant filter are not fully understood by the author.

Adaptive Cancelers

The second alternative filtering technique is the use of multiple adaptive interference cancelers as notch filters. Reference on the use of adaptive interference

cancelers, and a general text on adaptive signal processing, is contained in Adaptive Signal Processing by Widrow and Stearns [10]. The application of adaptive interference cancelers as notch filters is best described through the use of a flow diagram of such a filter. A diagram showing a filter designed to cancel a single interference frequency is shown as Figure 4-26. This figure and basic explanation are contained in chapter 12 of Widrow and Stearns.

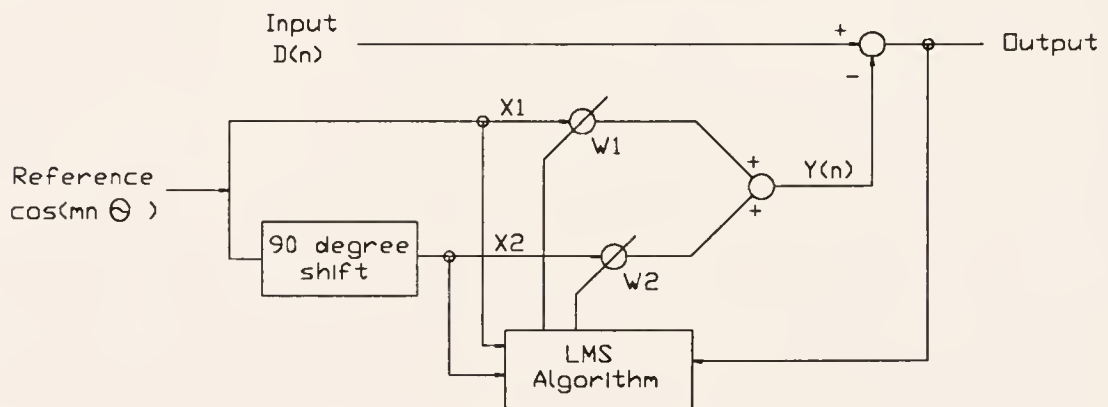


Figure 4-26 Signal Flow Diagram for Adaptive Notch Filter.

The primary input, $d(n)$, is the signal that is to have the interference filtered from it. The reference input is a signal that represents the interfering signal slated to be removed. In general this may be any interfering signal and is normally a measured signal. However, in the present case this reference signal would be the run-out frequency or one of its harmonics, and the primary input would be the measured sound signal. Because the reference signal is not available without containing the desired signal, the reference signal would be internally generated. That is, a digital reference signal for each harmonic that is to be canceled in the sound signal would be generated with their frequencies controlled by the measured speed of the spindle. The reference

signal and a 90 degree phase shifted reference signal are used, representing a sine-cosine pair at the desired frequency. The sine and cosine wave are each multiplied by an adaptive filter weight, w_1 and w_2 respectively, and then added together to form the output wave y . These weights are adaptively controlled by a least mean square adaptive algorithm. By changing the weights relative to each other, the phase of the output wave can be varied. By adjusting the weights in the same direction, the magnitude of the output wave can be varied. This output wave is then subtracted from the primary input signal forming the output of the filter ϵ . The least mean square algorithm adapts the weights to minimize the power in this output signal, thereby providing maximum cancellation of the reference input in the primary input. The formulation for up dating the filter weights is given by the following equations.

$$w_{1,k+1} = w_{1k} + 2 \mu \epsilon_k x_{1k}$$

$$w_{2,k+1} = w_{2k} + 2 \mu \epsilon_k x_{2k}$$

Where the subscript k is the sample number and μ is the adaptation constant. The diagram presented is the filter for a single harmonic. A similar filter would be used for each of the harmonics to be removed.

There are several concerns about the application of these cancelers to the filtering of the chatter signal. The first is the speed of adaptation. The adaptation speed is controlled by the constant μ . A large value allows rapid adaptation but introduces noise due to the adaptation process. Too large of a value of μ leads to an unstable filter. It is shown in reference [10] that the value of μ must be less than

one over the value of the maximum eigenvalue of the correlation matrix of the primary signal to insure stability. As a rule of thumb, the authors recommend that a value of less than one tenth of the trace of this matrix be used. Because there are multiple harmonics to be removed from the sound signal this leads to large eigenvalues due to the upper harmonics. Thus, in order to assure stability of the filters small values for μ will be required. This leads to a slow adapting filter. Further, the calculating of reference signals and then implementing a filter for each harmonic is computationally expensive. Indeed, the reference of each will have to be adjusted with the spindle speed to effectively filter during speed variations.

Synchronous Sampling

The final alternative filtering technique is to utilize the present comb filter or the proposed high-low pass and comb filtering technique on a synchronized sampled signal. This leads to simplifying advantages in the filtering but introduces other complications. Synchronized sampling is sampling the signal with a set number of samples per revolution of the spindle. A more complete explanation of this is contained in the tool breakage detection chapters of this report. The present synchronized sampling system receives its trigger signal from an encoder that is connected to the spindle. The encoder has limitations as to the maximum operating speed and cannot be used in a high speed system. This shortcoming may be avoided by using an electronic synchronized triggering system. This system would consist of a phase locked loop being driven by a pulse train from the spindle rotation. This pulse

may simply be a once per revolution edge, as used in the present system, from a reflectance type probe. A complete discussion of the use of phase lock loops and their use in frequency multiplication is contained in The Art of Electronics by Horowitz and Hill [11]. This system uses a reference frequency of digital edges and generates a signal that is at n times the reference frequency. The value of the counter, n , can be digitally controlled by the supervision computer. In this way a trigger signal of n times per revolution can be generated. Further, the number of samples per revolution can be controlled from the supervision computer.

Utilizing a synchronously sampled signal leads to advantages in the filtering just as it does in the tool breakage processing. Because the sampling is tied to the spindle speed the filter notches stay properly placed during spindle speed variations. The high-low pass filters are now designed in the per revolution domain (see tool breakage), instead of the time domain. In this way they are stationary in the per-revolution domain. That is, for a set number of run-out harmonics to be removed, the coefficients of the filters are constant with a varying spindle speed. Thus they only have to be calculated at start up instead of recalculated at each speed change as in the time based system.

The use of a synchronously sampled signal introduces some disadvantages to the processing. Because the signal is revolution-based instead of time based, the fourier transform of the signal is now in the per-revolution domain instead of the frequency domain. The chatter signal is at a set frequency, thus if the spindle speed is varying during chatter, the chatter is not stationary in the per revolution domain. Thus

the chatter signal will be blurred over several lines in the transformed domain. The extent of this blurring will depend on the level of spindle speed variation. How strongly this will effect the chatter detection would have to be investigated. Of course, when chatter is detected its frequency would be known in the cycles per-revolution domain and the spindle speed during this detection must be known in order to calculate the new command speed for the spindle in RPM.

In setting up a synchronous sampling system, consideration would have to be given to the fact that the band width of the detection system now changes with spindle speed. That is the Nyquist frequency is now expressed as,

$$f_{nyquist} = \frac{N * Spd}{120}$$

where N is the number of samples per revolution and Spd is the spindle speed in revolutions per minute. Thus N must set high enough to have sufficient band width at low speed and low enough not to overload the computations or data acquisition hardware at the maximum spindle speed. The value may be changed for different spindle speed ranges or the data down sampled if the range of spindle speed warrants. A single analog anti-aliasing filter may be used with the system. The cutoff frequency of the filter should be set just above the maximum frequency of interest of the chatter detection system. The number of samples per revolution should then be set such that, at the lowest spindle speed the nyquist frequency is not less than the cutoff frequency of the filter.

CHAPTER 5

TOOL BREAKAGE INTRODUCTION

Digital filtering techniques are applied to the area of tool breakage. It is desired to implement a system of detecting cutter breakage in process. This system should be responsive to slowly developing cutter damage such as the progressive chipping of a cutter edge, and also responsive to catastrophic failure of an edge. The system is designed as two sub-systems to perform these two separate tasks. The sub-system to detect progressive cutter damage is configured to detect both progressive damage and the starting of the cutting process with a damaged edge. This system will not detect cutter wear, unless that wear is isolated to a single tooth. Further, the sub-system is designed to reject the force or displacement signal due to cutter run-out. The second sub-system is designed to detect sudden failure of a cutting edge and quickly stop the cutting process in order to prevent further damage to the cutter, work piece or machine. By separating these two tasks, it is possible to utilize approaches that have been previously implemented by others. The previous implementations have suffered from being required to detect both failure modes, and this requirement prevented either system from being optimized for its specific task.

Both tool breakage sub-systems operate on a synchronously sampled displacement signal from the machine tool spindle. The parameters that describe the cutting process and the form of the breakage signal are common to both systems.

Hence the common features will be discussed first and then the individual systems and their performance will be described. The background of each subsystem, and their previous implementation will be covered with the individual discussions.

The task of the filtering in a cutter breakage detection system is to remove, from the cutting signal, the content that is due to normal cutting conditions. In this way the remaining signal is determined to be due to abnormal conditions such as cutter breakage. Content due to other abnormal conditions, such as chatter, may also be present. These signals may cause the cutter breakage system to false trigger. These issues will not be discussed here. Instead, the discussion will start with a model of the cutter and cutting process and in this way the sources and form of the signal content can be explained. Once the signal content is understood, with both a broken and unbroken cutter, the filtering to expose the cutter breakage feature will be covered.

Synchronized Sampling and the Per Revolution Domain

The signals used in the detection of cutter breakage are all synchronously sampled. That is they are sampled in synchronization with the revolution of the spindle. This technique was utilized by Altintas, Yellowley and Tlustý [12], and leads to great simplification of the processing of the signal. The synchronous sampling is accomplished by using an encoder attached to the spindle. In the present configuration the encoder produces two signals. One signal is a once per revolution pulse and the other is a pulse train of 120 lines equally spaced around the revolution of the spindle. By utilizing the 120 pulses per revolution as a trigger signal to acquire

the samples, data is collected at the same points during each revolution of the spindle. One hundred and twenty lines are used because it is integrally divisible by eight, six, four, three and two. This means that for cutters with these numbers of teeth, there will be an integer number of samples collected during each tooth period.

In discussion of the application and design of digital filters, it is convenient to look at the response of the filter and its effects on the processed signals in the frequency domain. When processing time-based signals, the signals are fourier transformed into the frequency domain and the impulse response of the filter can also be transformed directly into the frequency domain. However, the synchronously sampled signal is not a time-based signal but a rotation based signal. The fourier transform of these rotation based signals casts the signal in a per revolution domain as opposed to the frequency domain. Similarly the impulse response of filters based in the rotation domain can be projected into the per revolution domain.

For further illustration, the discrete fourier transform of a time based signal may be written,

$$X(k\Delta f) = \sum_{n=0}^{N-1} x(nT) e^{j(\frac{2\pi}{N})(-nk)}, k=0, 1, \dots, N-1$$

where,

N is the length of the signal in points

T is the time step, (1/sample frequency)

Therefore, $x(nT)$ represents the sampled time signal and $X(k\Delta f)$ is the transformed frequency series, where Δf is the frequency spacing and equal to sampling frequency divided by the number of points, N . In the rotation base system the sampling frequency is in samples per revolution as opposed to samples per second or hertz. Specifically, our sampling frequency is 120 samples per revolution. The samples are located at $1/120$ revolution separation and the resolution of the fourier transformed series is $120/N$ cycles per revolution. The Nyquist criteria dictates that the maximum resolved signal is not above 60 cycles per revolution.

Obviously, for a constant spindle speed the per-revolution domain can be directly mapped into the frequency domain by simply scaling the per revolution axis. The scaling factor will be different for each speed and vary during spindle speed variation. The advantage of the synchronously sampled signal is that the filters designed in the per revolution domain have constant coefficients even with non constant spindle speed. In contrast, filters used on a time based series which must have their coefficients recalculated to adapt to a changing spindle speed.

Throughout the following discussions of the cutter breakage detection schemes, the per revolution response of the filters will be shown along with the content of the cutting signal. It should be realized that these are used for illustration of the processing and that the signal is never transformed into this domain. All processing and detection is carried out in the time, or more properly, the revolution domain.

Model of the Cutter and Signal due to Tooth Position

A figure of a typical inserted cutter is shown in Figure 5-1. The main cutter body supports individual cutting inserts that may be individually indexed or replaced. All of the discussion pertaining to cutter breakage will be restricted to cutters with integrally spaced teeth (when the space between each sequential tooth pair is the same as all others on the cutter). Either type of cutter, the indexable or the integral cutter, is normally mounted to a tool holder that is held in the spindle taper.

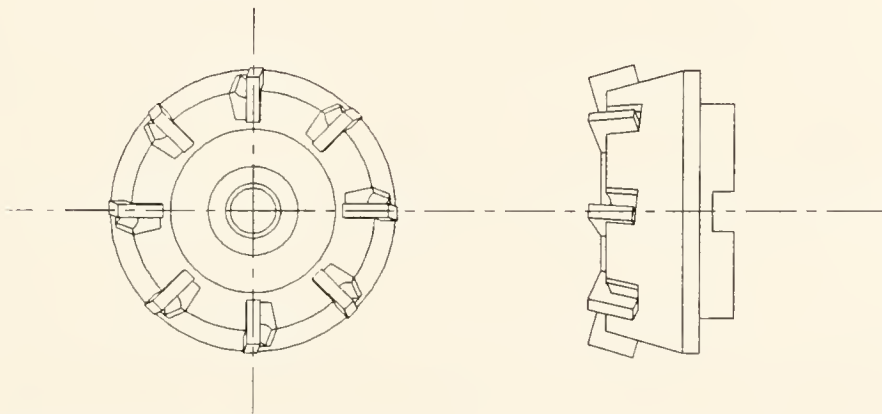


Figure 5-1 Replacable Insert Face Mill.

The basic cutting parameters and the terms used to describe them will now be discussed, including errors of tooth position and their effects on the cutting force signal. Axial immersion is defined as the axial depth of the cut, the depth of the cut along the direction of the rotation axis of the spindle. The radial immersion is the immersion perpendicular to the axis of rotation. Radial immersion is often expressed as a percentage of the cutter diameter, and the term slotting refers to 100 % radial

immersion. The chip load is the distance of travel of the center of the cutter per tooth period. A diagram depicting these parameters is shown as Figure 5-2.

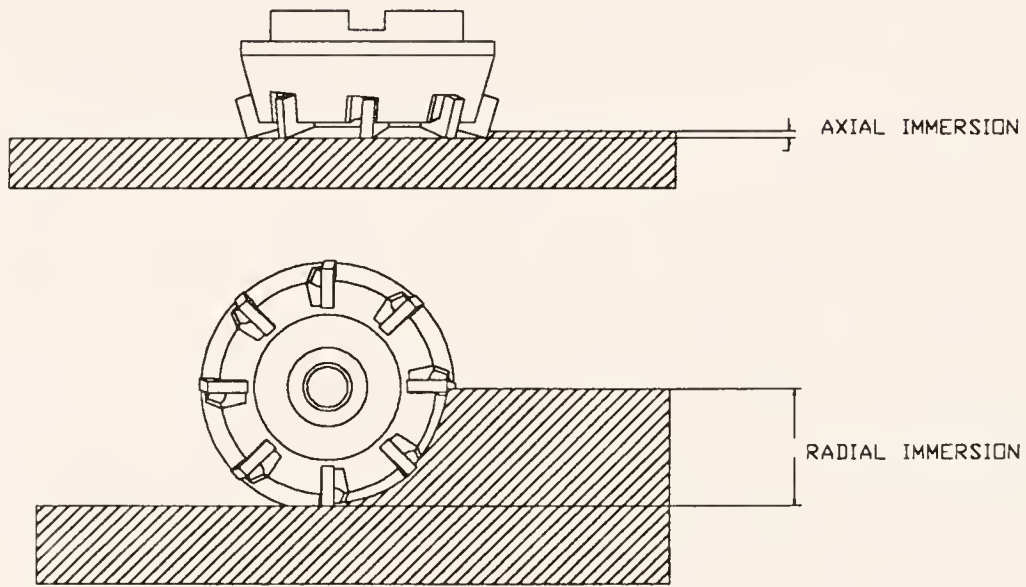


Figure 5-2 Radial and Axial Immersion.

The mode of milling is defined by the relative position of the work piece to the direction of the cutter path and its rotation. Up milling is the case where the cutting teeth enter the work piece with zero or small chip thickness and exit with a larger thickness. How much larger is dictated by the chip load and the radial immersion. In down milling, the teeth enter the work piece with some chip thickness and exit with a reduced or zero chip thickness. The slotting case can be defined as neither up nor down milling, as the teeth enter and exit the work piece with zero chip thickness. Up milling, down milling and slotting are diagramed in Figure 5-3.

Ideally, the teeth of a milling cutter are exactly located on a flat circle and this circle of teeth rotate about the center on the spindle axis. In reality, there is error in

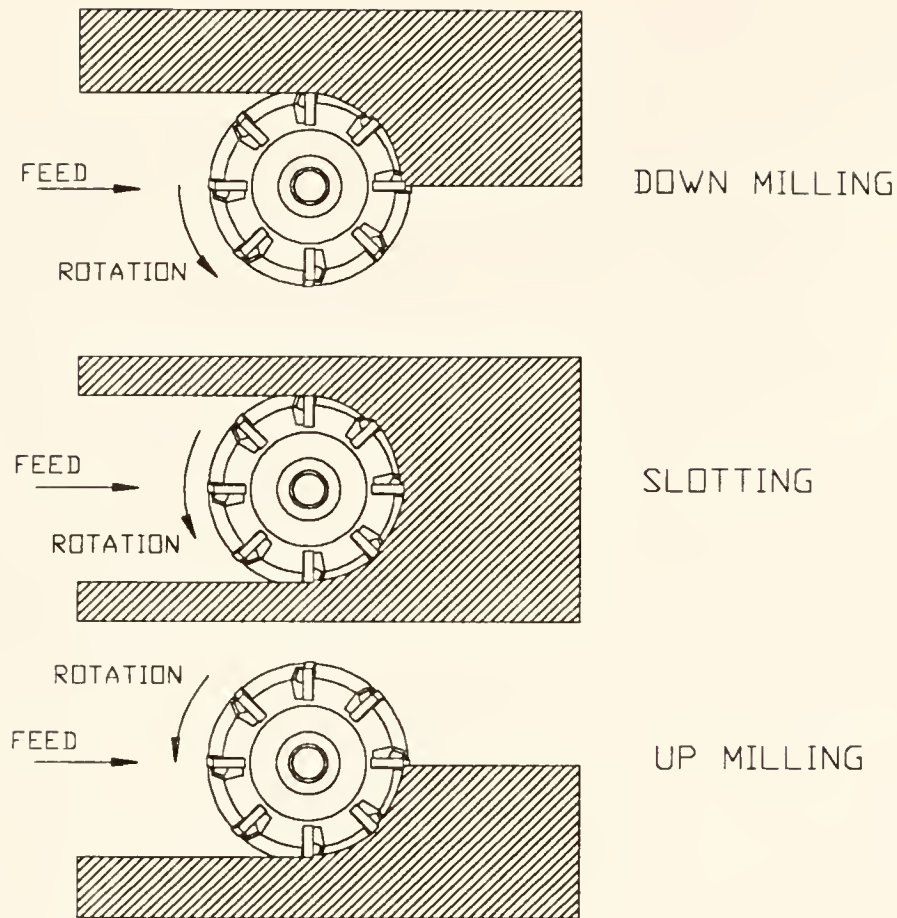


Figure 5-3 Milling Mode Diagram.

the location of the milling teeth. There are two main errors of position of the teeth that effect the cutting signal for the detection of cutter breakage. There may be run-out of the cutter. Run-out is the result of the center of the circle not being located at the center of rotation of the spindle. Run-out can also be produced by the plane of the circle not being normal to the axis of rotation of the spindle. Both of these conditions lead to a varying force signal that has strong content at the once per revolution frequency. Because the run-out produces a harmonic variation, this content is isolated to the once per revolution frequency. A figure diagraming these errors is shown as

Figure 5-4. As the effect of either form of run-out is the same for the cutter breakage schemes, the two will not be differentiated in further discussion. The second main error of position is that of tooth throw. Radial tooth throw is the misplacement of the teeth off the ideal circle, while axial throw is the misplacement of the teeth out of the plane of the circle. The signal due to either type of tooth throw is periodic in nature but not harmonic. This is because the signal content is spread on the once per revolution frequency and its harmonics. It will be shown later that this is the same type signal generated from a broken cutter tooth. In fact, the difference in tooth throw and cutter breakage is only a matter of magnitude. A severe tooth throw will have the same effect as a broken cutter on the cutter breakage detection system, as well as its effect on the actual milling operation. Radial and axial throw will be collectively referred to as tooth throw and are depicted in Figure 5-5.

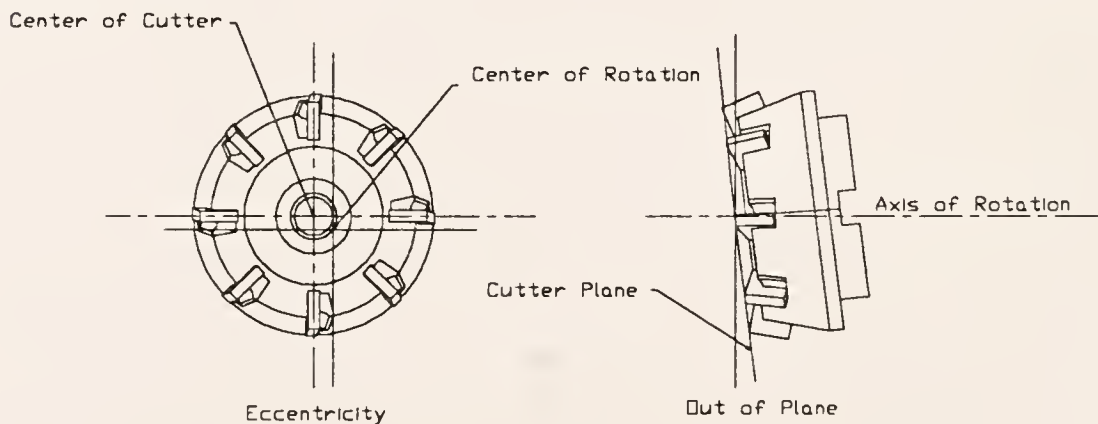


Figure 5-4 Cutter Run-out due to Eccentricity and Out of Plane Conditions.

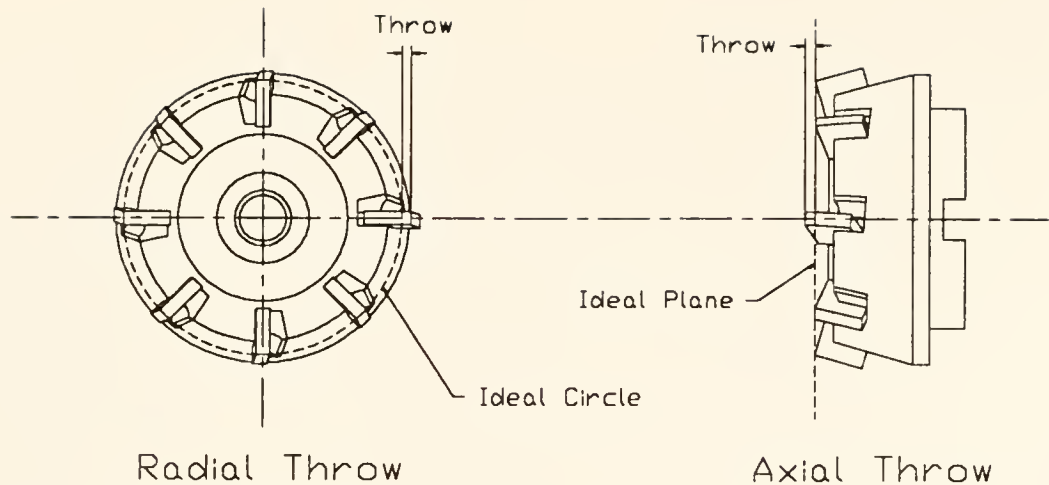


Figure 5-5 Diagram of Radial and Axial Tooth Throw.

Another error in the collected signal which is not truly present in the cutting force is the error of 'instrument run-out'. In its present form, the cutter breakage detection system, as implemented at the Machine Tool Research Center at the University of Florida, utilizes the displacement of the spindle relative to the machine tool frame as representative of the cutting force. This displacement is measured by using inductive eddy current probes mounted on the spindle housing, which measure the gap between themselves and a target ring mounted on the spindle. This signal is not the true cutting force signal but the force signal filtered through the displacement versus force transfer function of the spindle tool combination. The instrument run-out arises from the target ring not being truly round or centered on the spindle. As the spindle rotates, any eccentricity of the ring produces a harmonic displacement signal at the once per revolution frequency. Additionally, any unroundness produces higher frequency content in the displacement signal. The signal due to these errors is ideally filtered by measuring the run-out profile of the ring during idle of the machine. This

run-out profile is then subtracted from the measured data at each revolution. In this way the instrument run-out is removed from the measured data. For all future discussion the term run-out will refer to actual cutter run-out as opposed to instrument run-out.

The Cutting Force Signal and Milling Simulation

The modeled cutting force signal used in this study is based on the 'Instantaneous Rigid Force Model' as presented by Smith and Tlustý in [13]. This model assumes the cutter to be rigid and hence, does not allow cutter displacement to participate in the evaluation of the cutting forces. In this way no regenerative displacement is considered. When regenerative effects or 'dynamics' are included, it will be noted. In the instantaneous rigid force model, the tangential force is considered proportional to the area of the chip being removed. The tangential force, F_t , is proportional to the chip width b , the instantaneous chip thickness f , and the cutting stiffness of the work piece material K_s .

$$F_t = bfK_s$$

The Radial force is assume to be proportional to the tangential force.

$$F_r = 0.3 F_t$$

The total force on the cutter is the vector sum of the individual force on the teeth.

The simulation used in this work was written by Chris Vierck [14] and based on a simulation routine written by Carlos Zamudio. The simulation allows for various radial and axial immersions, spindle speeds and tooth numbers. Simulations can be carried out for steady state and entry and exiting conditions. Provisions are available to simulate sudden cutter breakage, slow chipping and to, include cutter run-out, tooth throws and random noise content in the signal. A complete description and listing of the program can be referenced in Mr. Vierck's Thesis [14].

The Cutting Force Signal

In order to understand the cutter breakage detection schemes it is first necessary to understand the form of the cutting signal itself. In its present implementation, the cutter breakage detection system utilizes displacement signals from the spindle in two orthogonal directions. The cutter breakage system works on the overall force or displacement of the spindle, hence the vector sum of the two directions is used by the detection system. Because of this, the remaining work will refer to the force or displacement with the understanding that this is the vector sum of the two directions.

Figure 5-6A shows a picture of the simulated cutting signal for an ideal unbroken milling cutter. The cut is of shallow immersion with only one tooth engaged with the workpiece at a time. As each tooth engages, the force rises as the chip width increases and then drops as the tooth exits the work piece. The signal is seen to be perfectly periodic in that each tooth is identical to the others. The fourier transform of

this force signal is shown in Figure 5-6B. The cutter is an eight tooth cutter for this case. The signal contains spectral content only at the steady state, the tooth frequency (eight cycles per revolution) and its harmonics.

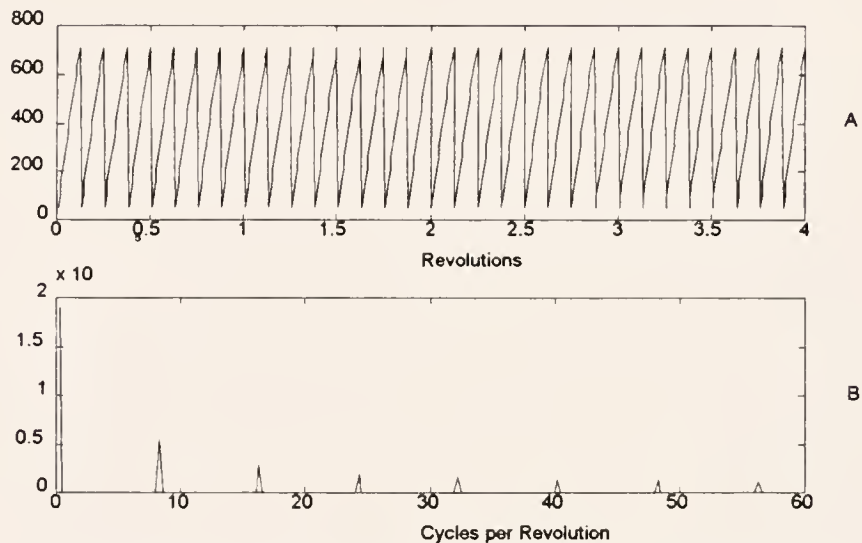


Figure 5-6 Simulated Cutting Force Signal for an Eight Insert Cutter and the Per Revolution Domain Transform.

The signal simulated for an idealized broken cutter is shown in Figure 5-7A. For the idealized broken cutter, all of the teeth are considered identical except the broken tooth, which is completely absent. Because the broken tooth does not cut, the following tooth must cut twice as much material as normal, leading to an increased force spike for that tooth. The frequency content of this signal is shown in Figure 5-7B. It can be seen that there is now spectral content at the once per revolution frequency and its harmonics. The region below the tooth frequency, in this case eight cycles per revolution, has the most content and is referred to here as 'breakage region',

this term being borrowed from Tarng and Tlusty [15]. This is the content that is to be isolated in order to detect cutter breakage.

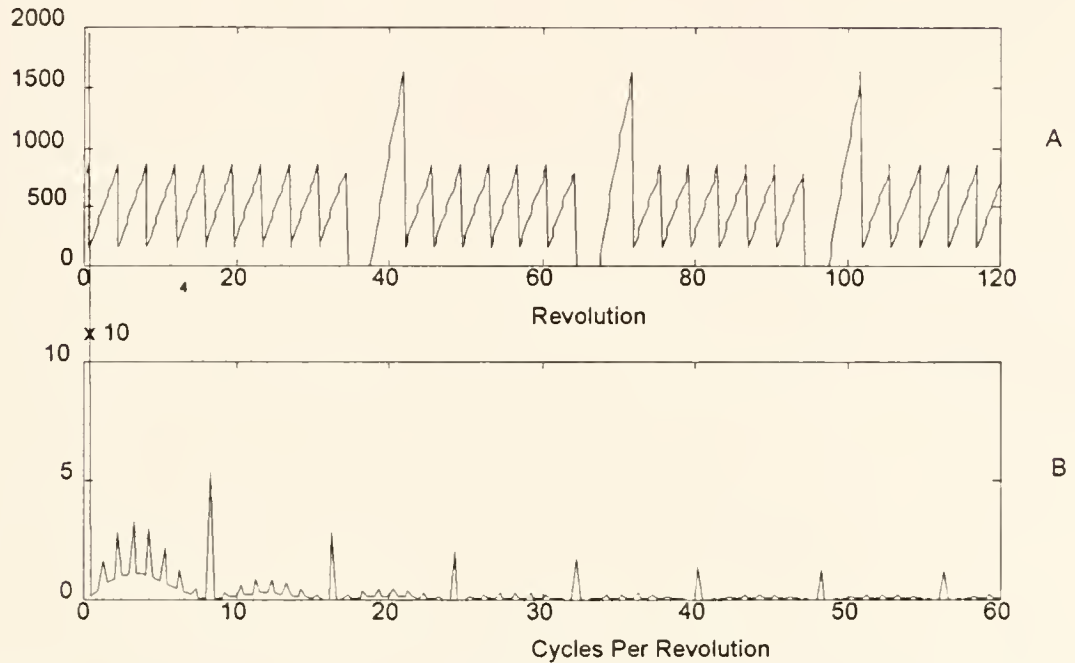


Figure 5-7 Simulated Cutting Force Signal for a Broken Eight Insert Cutter and the Per Revolution Domain Transform.

The effect of cutter run-out on the force signal can be seen in Figure 5-8A. Because the run-out is an offset of the center of rotation, the effect is to vary the cutting force at the run-out frequency. This is a pure harmonic variation so that all of the frequency content due to run-out is located at the once per revolution frequency. This can be seen in Figure 5-8B. The effect of tooth throw is very different than that of run-out, as can be seen in Figure 5-9A. As each tooth varies slightly in its position, the signal is periodic but not harmonic at the rotation frequency. This leads to frequency content, as shown in Figure 5-9B, at the rotation frequency and its harmonics. As was stated previously, this signal is only differentiable from the broken

cutter signal by the scale. As the effects of tooth throw and cutter breakage are the same, there is no reason to try and differentiate them. Thus, a severe tooth throw is simply considered tooth breakage. A further concern due to tooth throw is that the higher harmonics of the run-out may be significantly amplified by the spindle's transfer function and, due to the low sampling frequency, be aliased into the breakage signal. This will lead to significant content in the breakage region, even for a good cutter with minor tooth throws. This condition makes the separation of good and bad cutters very difficult, and will be addressed further in a later section.

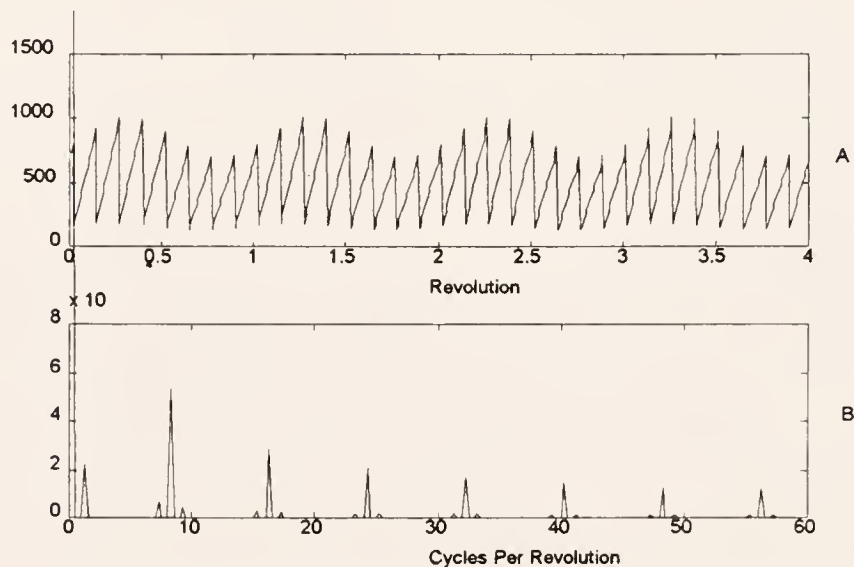


Figure 5-8 Simulated Cutting Force Signal for an Eight Insert Cutter with Runout and the Per Revolution Domain Transform.

The form of the cutting force signal changes with the radial immersion of the cutter. As more and more teeth are engaged in the cut at one time, their individual tooth forces average together. This somewhat obscures the signal due to the broken tooth. It should be noted that the signal content is always present in the breakage

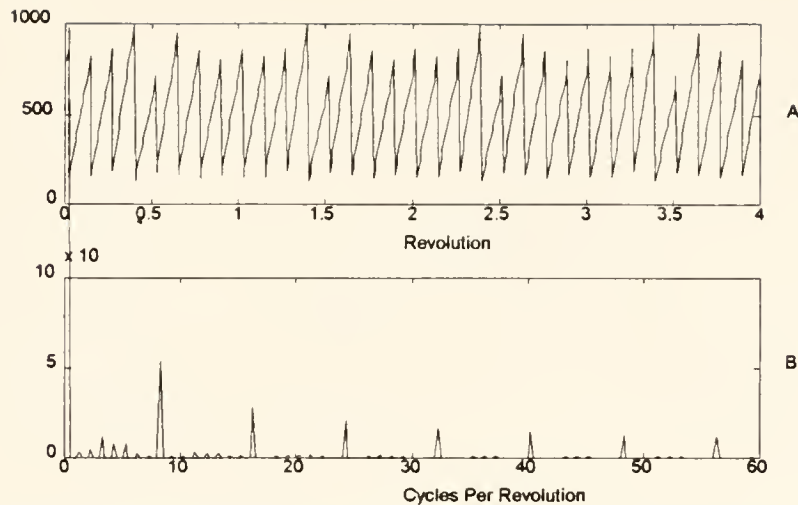


Figure 5-9 Simulated Cutting Force Signal for an Eight Insert Cutter with Tooth Throw and the Per Revolution Domain Transform.

region of the spectrum. The content in the higher harmonics is reduced as the broken tooth and the following tooth are engaged for a longer period with increasing radial immersion. Figures 5-10 and 5-11 show the simulated signal for an eight tooth cutter and the spectral content for 1/4, 1/2, 3/4 and full radial immersion. The figures show the signal for an unbroken and a broken cutter, respectively.

The transient milling cases of entry and exit are also of concern as the cutter breakage system must obviously handle these conditions. As the exit case is a mirror image of the entry case, only the entry case will be discussed here and it will be understood that the exit shows similar features. It can be seen in Figure 5-12 that as the cutter encounters the work piece, the teeth first enter into cutting near the maximum chip thickness. The duration of the engagement is shorter than during the later steady state cutting. This leads to a force spike that is essentially the same

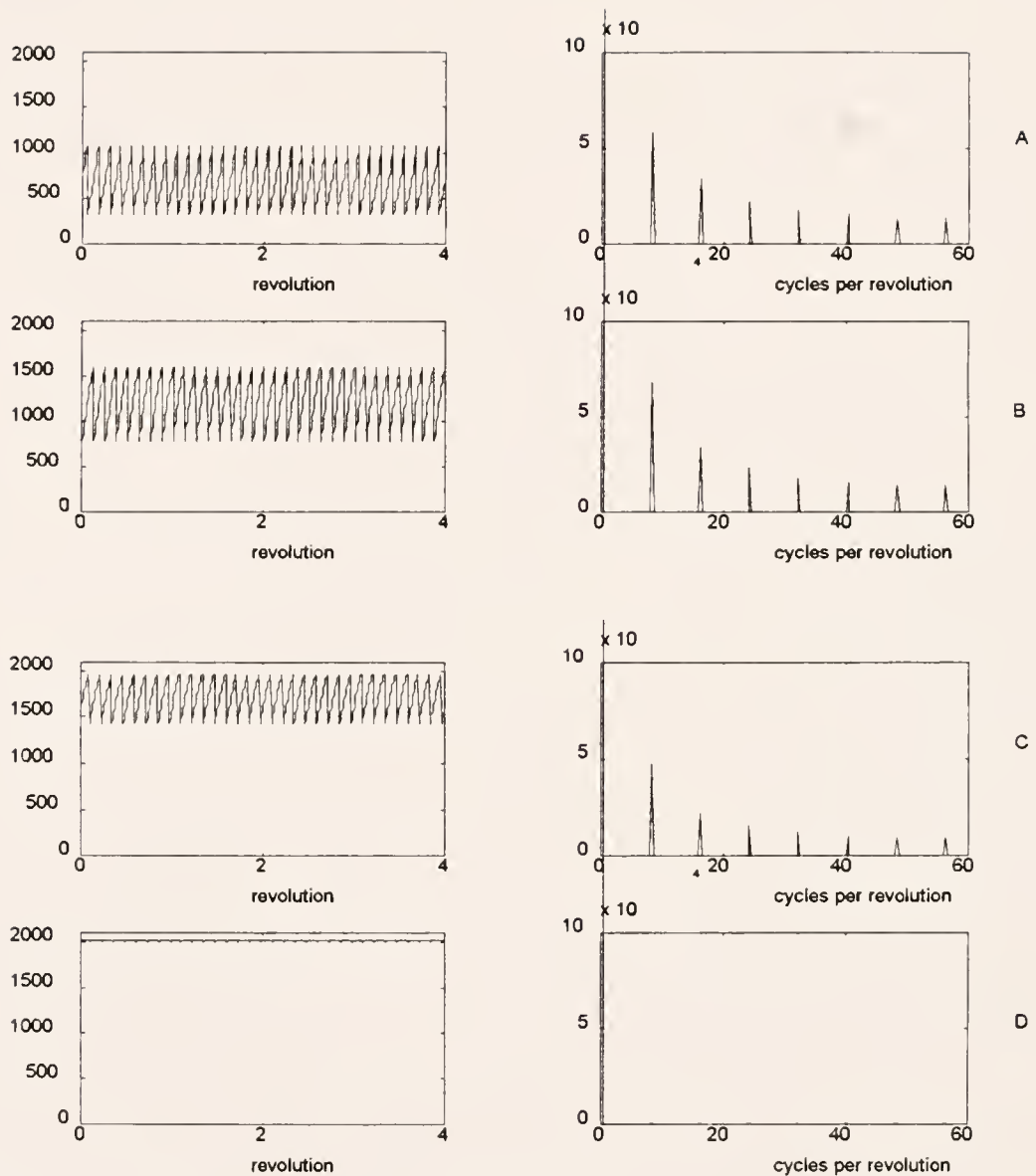


Figure 5-10 Simulated Cutting Force Signal and Frequency Content for an Unbroken Cutter, A) 25% B) 50% C) 75% D) 100% Radial Immersion.

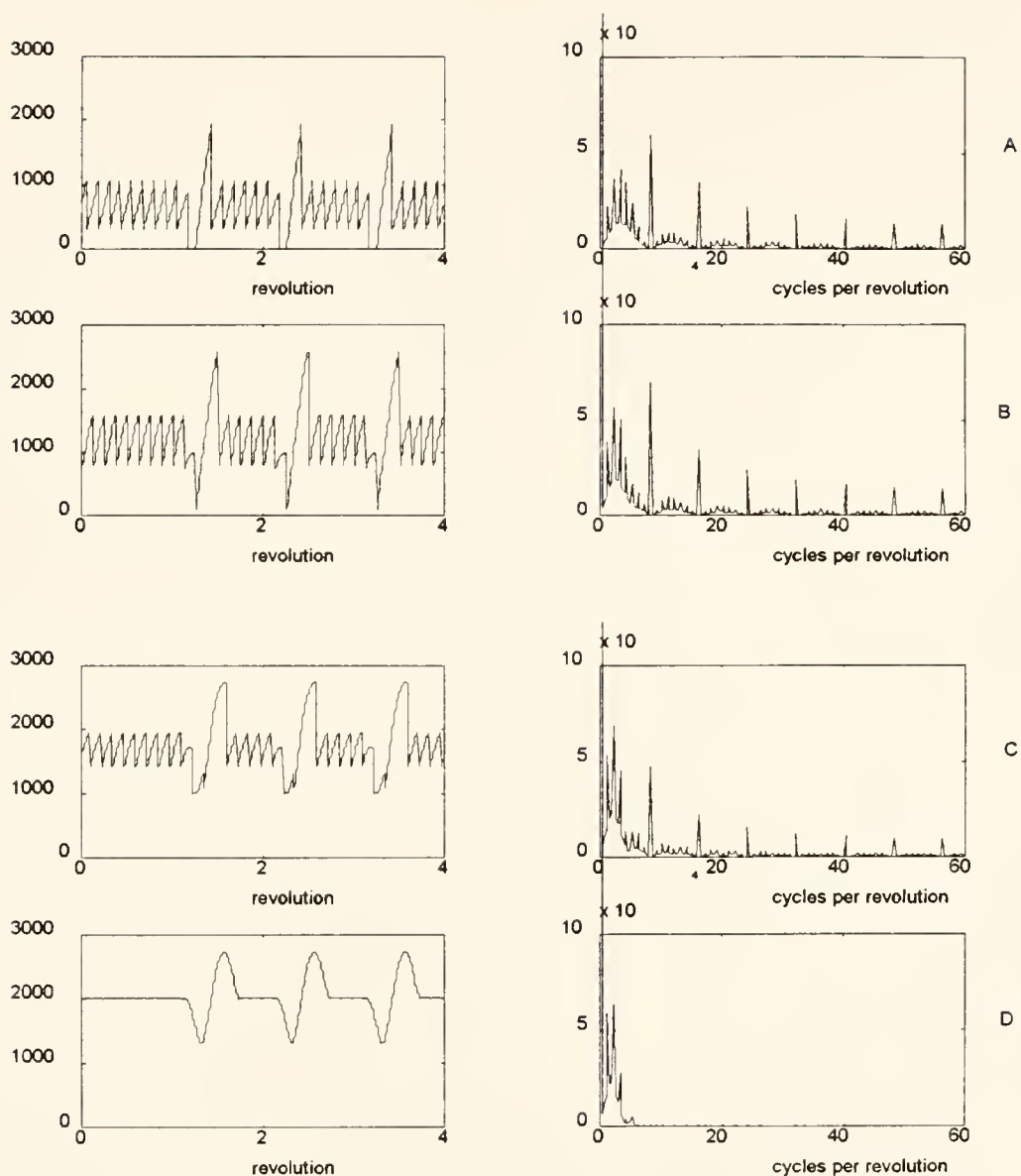


Figure 5-11 Simulated Force Signal and Frequency Content for a Broken Cutter, A) 25% B) 50% C) 75% D) 100% Radial Immersion.

height as the force signal during the steady state cutting, but is shorter in duration and located at the end of the individual tooth period. As the cutter becomes more and more engaged with the work piece, the pulse height maintains the same magnitude but the duration steadily increases, towards the first part of the tooth period, to its steady state value. This pattern is seen In Figure 5-13. This varying width pulse train requires averaging to avoid passing of the tooth frequency and its harmonics through other filters. This will be addressed in the individual sections on the cutter breakage sub systems.

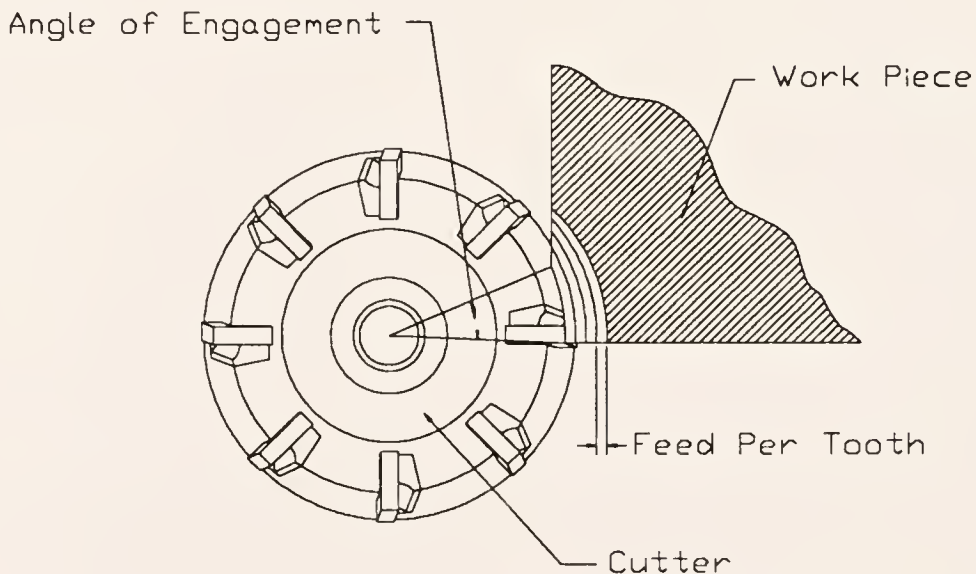


Figure 5-12 Diagram of Entry Cutting Condition.

A final cutting condition that must be considered is that of sudden entry. Sudden entry is the condition of the cutter entering the work piece such that all of the teeth engage at once. The sudden entry condition is encountered during cornering, as the cutter stops in the corner and then begins motion in the other direction, the cutter

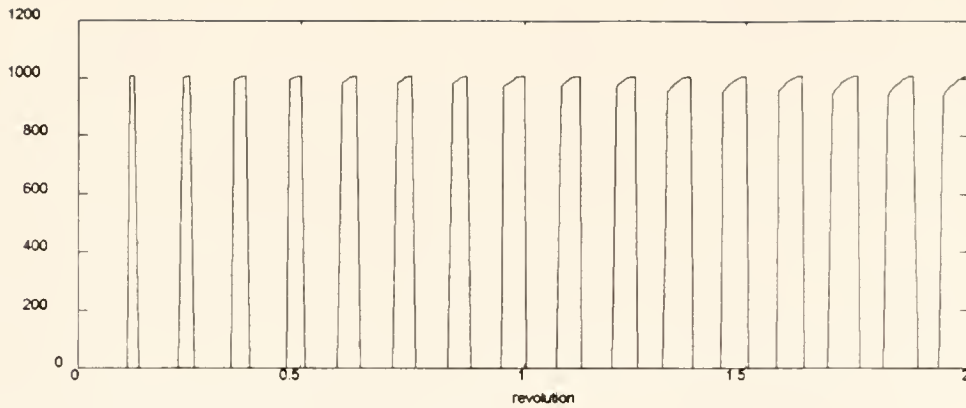


Figure 5-13 Simulated Entry Force Signal.

encounters a sudden entry with the start of the motion. The cutter breakage system must not false trigger on the sudden entry condition. The sudden exit is a similar condition at exit, when the cutter suddenly disengages. This condition is much less common than the sudden entry. The strategy to avoid triggering is the same as for sudden entry, therefore, this condition will not be covered.

CHAPTER 6

THE ONCE PER REVOLUTION DIFFERENCE SUB SYSTEM

The introduction stated that the cutter breakage system was broken into two sub-systems. One sub system is designed to detect catastrophic insert failure, and the other, slowly developing damage or previously existing damage. The specific design of the catastrophic failure sub system will now be covered, and the progressive damage sub system will be addressed in Chapter 7. The background and attributes of each sub system will be discussed in the respective chapters.

The first sub system addressed will be one based on a once per revolution filtering method. This sub system is designed to detect catastrophic insert failure during the cutting operation. The system is based on a once per revolution difference filtering method, that is the sample collected exactly one revolution back is subtracted from the present sample. Similar systems have been implemented in the past by Yoon, Principe [16] [17], O'Brien, Kapoor [18] and R. Walters [9].

These implementations vary slightly but are all based on a similar filtering scheme. The once per revolution difference filter is a comb filter with zeros placed at the run-out frequency and its harmonics. The transfer function in the per revolution domain is shown in Figure 6-1. The transfer function is only plotted in the range from zero to sixteen cycles per revolution for clarity of the figure. The form of the function

repeats throughout the frequency range to the Nyquist frequency. The difference equation for a system with 120 samples per revolution is written.

$$Y_n = X_n - X_{n-120}$$

Simply described, any signal that repeats once per revolution will be subtracted out of the filtered signal.

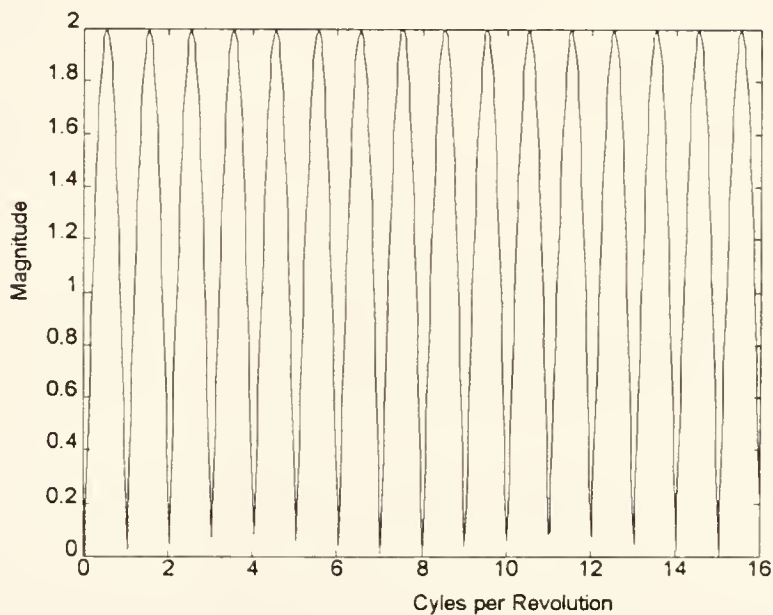


Figure 6-1 Once per Revolution Difference Transfer Function.

In steady state conditions, any signal content at the rotation frequency or its harmonics will be filtered out. The cutting signal from both a good or broken cutter has been shown to have content at only these frequencies. Therefore, the once per revolution filter filters out the entire signal due to a good or broken cutter during steady state conditions. The signal used to detect breakage is only present in a

temporal sense, that is there is only signal content when there is a change between revolutions. An illustration of this is shown in Figure 6-2A, where a simulation of a sudden cutter breakage is shown. The tooth is originally present and then completely breaks during cutting. The filtered signal is shown in Figure 6-2B. The signal due to the breakage is only present at the point of breakage, and after that point, the signal is completely filtered out.

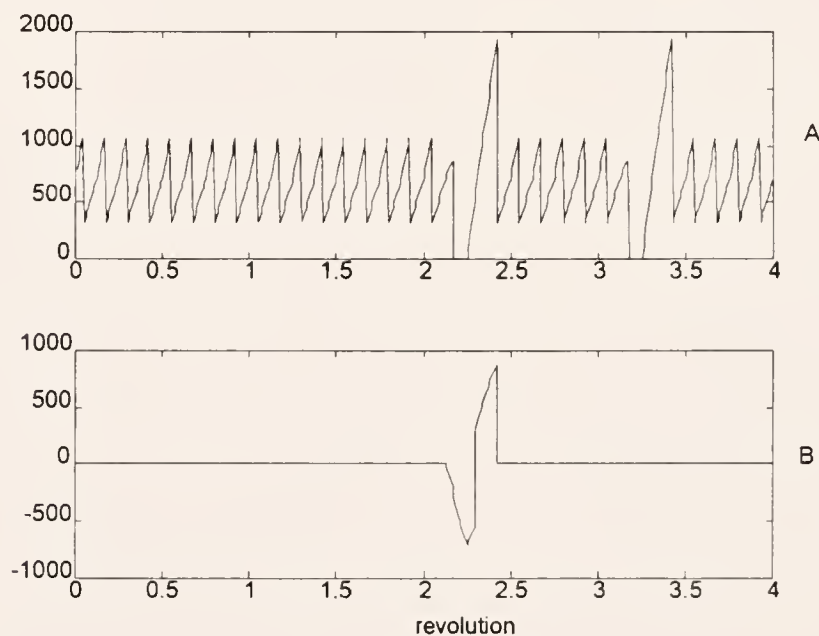


Figure 6-2 Simulated Sudden Breakage Signal for Eight Tooth Cutter. A) Raw Signal B) Filtered Through the Once Per Revolution Difference Filter.

Due to the fact that the once per revolution filter removes all the content of the cutting signal, and passes only the signal due to a sudden change, it is well suited to be used in the detection of catastrophic tooth failure. Also, due to its complete filtering of the cutting signal for both broken and unbroken cutters, the system is not sensitive to beginning the cutting process with a damaged cutter. The filtering can be

modified to give it some sensitivity to beginning with a damaged cutter as done by Yoon and Principe.

The system of Yoon performed a revolution difference with the average value from the several past revolutions. This filter can be written as follows:

$$Y_n = X_n - \left(\frac{1}{k}\right) \sum_{i=1}^k X_{n-120(k-i)}$$

where k is the number of previous revolutions used in the average. In steady state conditions this filter gives the same response as the simple once per revolution difference filter. However, during transient conditions such as entry and exit, some signal due to the broken tooth is passed through the filter. This can be explained by comparing the cutter signal to an average signal several rotations back. For the broken tooth, the zero force value of the average is always subtracted from the present zero force value. However, the following tooth has twice the force value, from which is subtracted the averaged force value from the previous revolutions. All other non-broken teeth have a single height force value from which is subtracted the appropriate previous average. In this way, the broken tooth shows a zero difference, while the following tooth shows a difference value twice that of the other unbroken teeth. This characteristic bi-directional signal can then be detected as indication of cutter breakage.

There are two other significant consequences of using an average of the previous revolutions. The first is that the breakage signal due to a sudden failure is now sustained in the signal for as many revolutions as were used in the averaging.

The signal level reduces with each revolution in proportion to the number used in the average, until all revolutions contain the broken signal and it is completely filtered.

The other effect is that due to comparing the cutting force to the force averaged over several revolutions, there is a 'slow wave' that is not filtered from the data during transient milling conditions. This slow wave was removed by Yoon by utilizing an additional median filter.

The point by point once per revolution difference filter completely filters the cutting signal during steady state conditions. This is not the case during the transient conditions such as entry. This can be seen by the form of the entry cutting force signal. The entry signal was previously shown to be essentially a stream of pulses of the same height and increasing duration with the increasing radial immersion. If a point by point difference is formed from this series with the one revolution back points, it can be seen that for each pulse, a pulse of the same height but less duration is removed. This leaves a pulse train that is the remaining portion of each pulse in the filtered signal. The resulting difference series is shown in Figure 6-3. Because the pulses are separated at the tooth frequency, it is obvious that the frequency content of this signal will be located at the tooth frequency and its harmonics.

To filter this remaining signal during the transient milling stages, O'Brien and Kapoor used an Autoregressive filter with adapting coefficients to model the data. The model was based on the past three revolutions of data and was formed with delays such that the tooth frequency and its harmonics could be matched with relatively few coefficients. Specifically, for a four tooth cutter and using 60 samples per revolution,

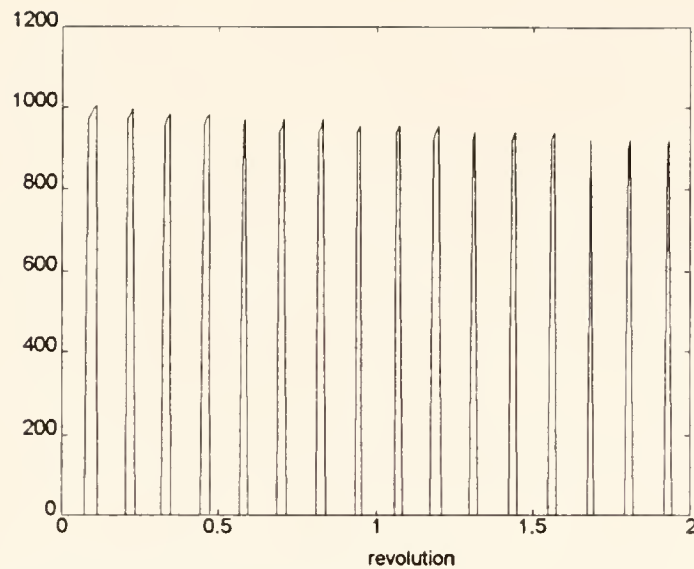


Figure 6-3 Entry Signal Filtered Through the Once Per Revolution Difference Filter.

delays of 15, 30, 45 and 60 were used. Delays of one and two were also included to model the slowly varying component of the signal. From this model, the next sample value was predicted. This predicted value was then subtracted from the actual sampled value forming a 'forecast error series'. The forecast error series was then mostly filtered of the transient conditions. Because the model could not adapt rapidly enough to model a sudden tooth failure, tooth failure was passed through to the forecast error series. The forecast error series was bound by thresholds based on the statistical variation of this series. Cutter breakage was determined to have occurred when the forecast error series exceeded either the positive or negative threshold.

Although the method of O'Brien and Kapoor was successful, a much simpler method of treating the transient case is proposed. By applying an averaging filter with the average length equal to the tooth period, these problems can be avoided. The averaging spreads the sharp pulses out over the entire tooth period, thereby greatly

attenuating the frequency content at the tooth frequency and its harmonics. When the once per revolution filter is then applied, only a small slow wave remains in the signal. The averaged entry signal and the once per revolution filtered signal are seen in Figure 6-4. For comparison, the per revolution domain transform of both the non-averaged and the averaged entry cases are shown in Figure 6-5. The difference equation for the averaging filter can be written as follows:

$$Y_n = \left(\frac{m}{120}\right) \sum_{i=0}^{\frac{120}{m}-1} X_{n-i}$$

where m is the number of teeth on the cutter.

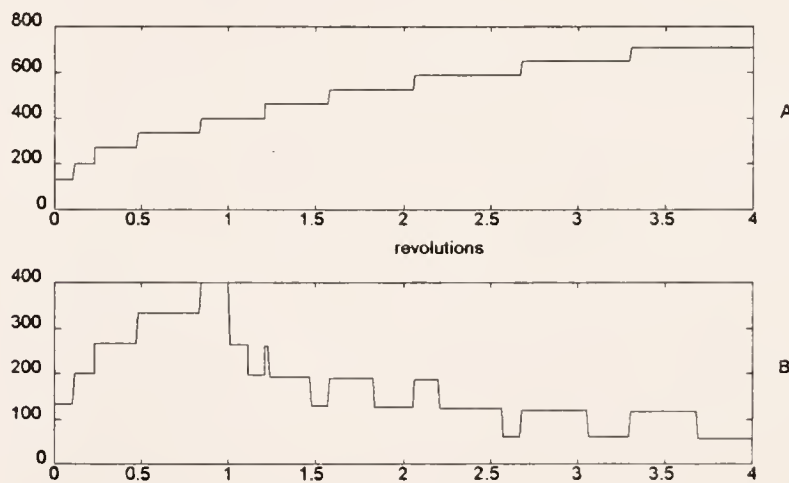


Figure 6-4 Simulated Entry Signal A) Filtered Through The Running Average Filter and B) The Averaged Signal Filtered Through the Once Per Revolution Difference Filter.

The proposed sub system will consist of first passing the cutting signal through an averaging filter and then through a point-by-point once per revolution difference filter. The filtering will remove all content at the run-out frequency and its harmonics, including all the signal from the cutting force signal. The only signal content

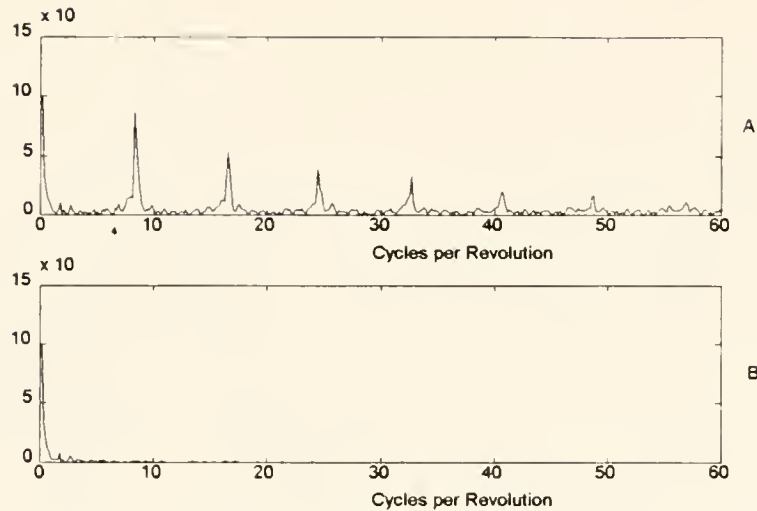


Figure 6-5 Per Revolution Content of the Entry Signal. A) Non-Averaged Signal B) Averaged Signal.

remaining will be noise and a slight slow wave during transient cutting conditions.

The sub system is thus completely insensitive to run-out of the cutter or beginning a cut with a damaged cutter. The slow wave is removed from the signal by subtracting the DC offset of the signal. This offset is estimated as the average value of the data points for the previous revolution.

The only content remaining in the filtered signal is due to insert failure, noise, or a sudden change in the cutting force pattern as introduced from a sudden entry. The breakage signal is only present at the point of catastrophic tooth failure and is of the form of a bi-directional peak. As the force value from this bi-directional peak is substantially greater than the noise content of the signal, thresholding the signal is relatively easy. The force signal due to a sudden entry is one directional. Therefore, to avoid a false trigger in sudden entry, it is only necessary to require that the signal break both a positive and negative threshold.

The thresholds for the system are based on a multiple of the remaining noise content in the signal. This noise level is based on the average of the absolute value of the filtered sample values for three revolutions. The thresholds are based on three revolutions of noise to assure that they are not strongly effected by a few high points in the data stream. Further, the three revolutions used are delayed by one revolution before comparing them to the filtered signal. This delay is used to assure that high values in the filtered signal, from cutter breakage, do not immediatly affect the threshold level and thereby, possibly rapidly increasing the threshold and preventing triggering during the breakage. The threshold values can thus be expressed as follows:

$$Thr(n) = \pm C * \frac{1}{360} \sum_{i=120}^{480} |X_{n-i}|$$

The filtered signal is directly compared to the threshold limits to determine cutter breakage. If the filtered signal values are divided by the estimate of the noise level, the output may simply be compared to the positive and negative of the multiplier C.

An overall flow diagram of the detection sub system is shown in Figure 6-6.

The input signal is assumed to have already been filtered of the instrument run-out.

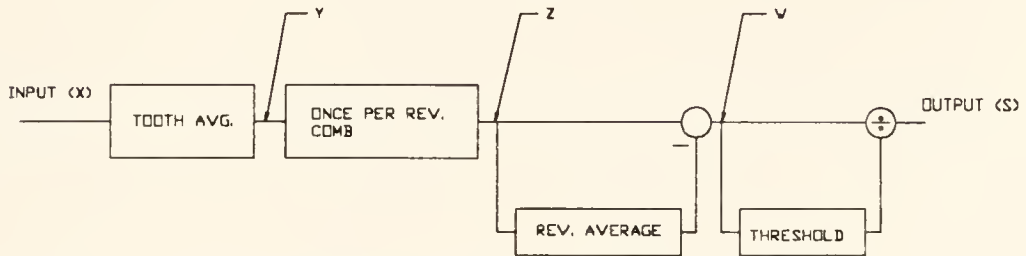


Figure 6-6 Flow Diagram of the Sudden Breakage Detection System.

Where each of the variables can be defined as follows: the tooth average,

$$Y(n) = \frac{m}{120} * \sum_{i=0}^{\frac{120}{m}-1} X(n-i)$$

where m is the number of teeth on the cutter, the once per revolution difference,

$$Z(n) = Y(n) - Y(n-120)$$

the revolution average,

$$W(n) = Z(n) - \frac{1}{120} * \sum_{i=0}^{119} Z(n-i)$$

and the threshold.

$$S(n) = \frac{W(n)}{\frac{1}{360} * \sum_{i=120}^{479} |W_{n-i}|}$$

The value of $S(n)$ is therefore directly compared to a constant, as determined by the operator.

The resistance of the filtering scheme to cutter run-out and tooth throws is demonstrated in Figure 6-7. The Figure shows a simulated cutting signal for an eight tooth cutter with 50% radial immersion. The cutter is modeled as having 25% cutter run-out and 25% tooth throw. Where the percentage run-out is defined as the amplitude of the run-out as a percentage of the feed per tooth, and the throw is randomly distributed over the teeth, with the maximum throw equal to the specified percentage of the feed per tooth. The simulated cut is of steady state cutting with a sudden breakage of the cutter on the seventh revolution. Five percent random noise is also added to the signal. The system is seen to completely filter the cutter run-out and tooth throw from the cutting force signal, while clearly passing the cutter breakage signal.

The effects of the averaging filter on the entry transient are demonstrated in Figures 6-8 through 6-11. Figure 6-8 shows the vector sum of a displacement signal from an eight tooth cutter with the teeth set with in 0.001 inches of tolerance. The data is from cutting tests on a Sundstrand series twenty Omnimil at the University of Florida Machine Tool Research Center. The Omnimil is equipped with inductance displacement sensors in both the X and Y directions. The plotted signal is from a 600 RPM cut in the X direction, with 25% radial immersion. The chip load is 0.008

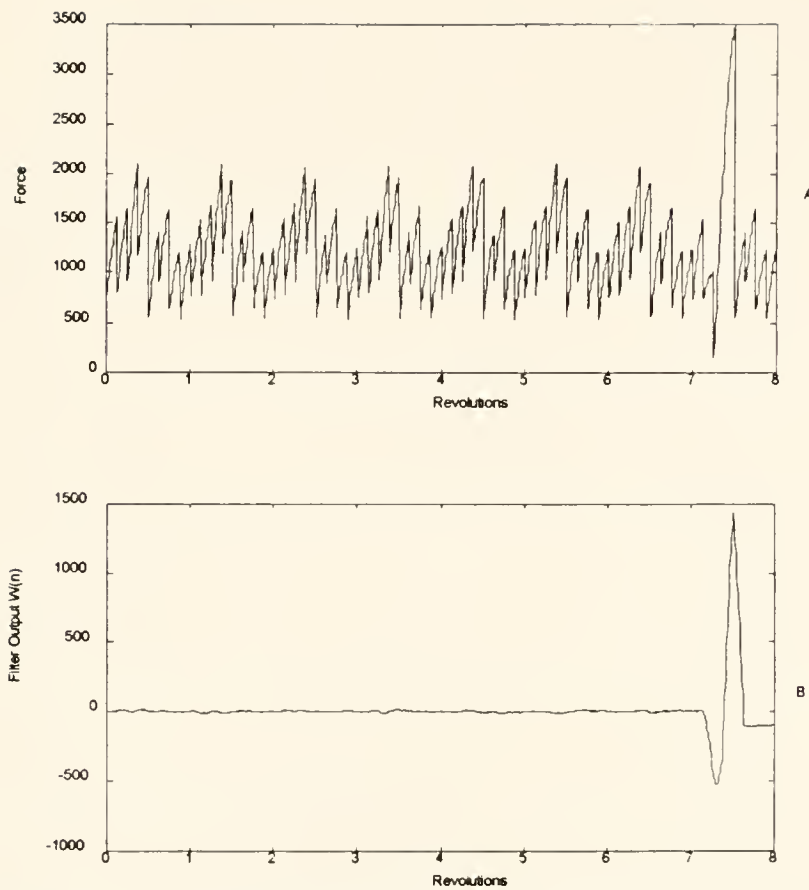


Figure 6-7 Simulated Cutting Force Signal with Breakage for an Eight Tooth Cutter with 25% Run-Out, 25% Tooth Throw, and 5% Random Noise. A) Force Signal B) Signal Filtered Through the Once Per Revolution Sub System.

inches per tooth and the work piece material is cast iron. The instrument run-out has already been removed from the plotted data. The entry transient of the cut is plotted in the figure.

Figure 6-9 shows the entry transient filtered through a point by point revolution difference filter. The filtered data still contains strong tooth frequency content and its harmonics. The spectrum of the filtered entry transient is shown in Figure 6-10. This spectrum is of revolutions six through twelve of the displayed data.

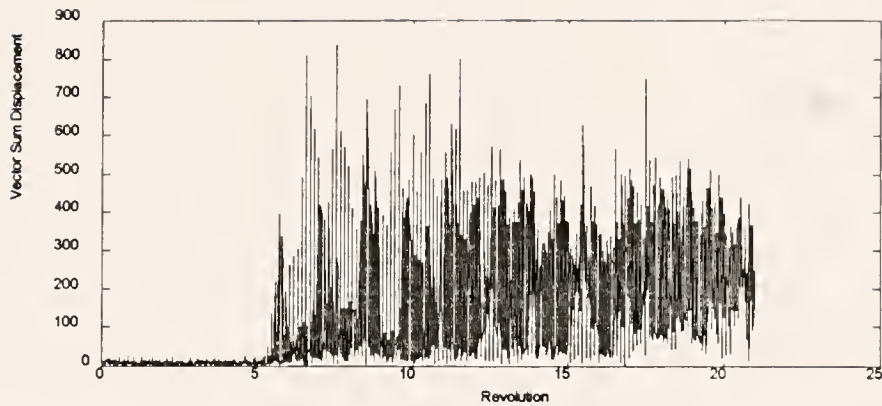


Figure 6-8 Vector Sum Displacement Signal for Eight Tooth Face Mill, 25% Radial Immersion, Entry Transient.

Figure 6-11 shows the entry transient filtered through both the averaging filter, with a window length equal to the tooth period, and the point-by-point revolution difference filter. The filtered signal is seen to be rid of the tooth frequency noise and only contains a slight slow wave during the transient entry.

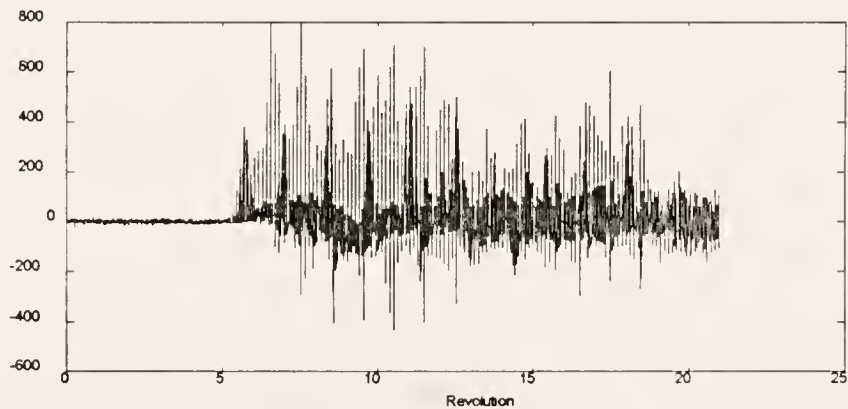


Figure 6-9 The Entry Transient Displacement Signal, Filtered Through a Point-by-Point Per Revolution Filter.

In order to test the sudden breakage sub system, it is necessary to have cutting data in which the cutter fails in process. To facilitate these controlled breakages of the cutter, inserts were modified so that they could be made to fail during cutting tests. A diagram of the modified insert is shown in Figure 6-12. The modified inserts were titanium nitride coated carbide and were used for cutting a steel work piece. The inserts are modified by cutting a 0.10 inch deep groove diagonally across the corner of the insert using a 0.035 inch thick diamond coated cut off wheel. The

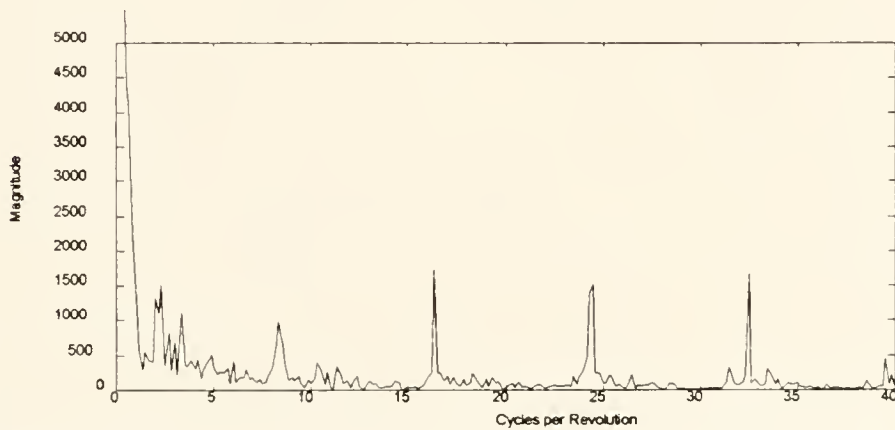


Figure 6-10 Spectrum of the Point-by-Point Difference Signal for an Entry Transient.



Figure 6-11 Entry Transient Filtered Through the Averaging Filter and the Point-by-Point Revolution Difference Filter.

opposite side of the insert is also relieved with the diamond wheel to remove the support for the corner by the insert holding nest. A modified insert is loaded in a face mill with the other inserts unmodified. The tool is then used in cutting. During the cut, the feed rate override is used to progressively increase the feed and therefore the chip load until the insert fails in process.

A typical record of the raw data from an in process breakage is shown in Figure 6-13. In this record, the modified insert had been shimmed out in the radial direction to increase its chip load relative to the other teeth on the cutter, representing

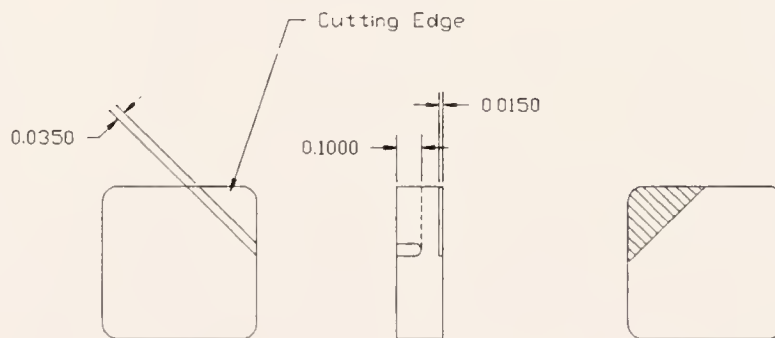


Figure 6-12 Insert Modifications for in Process Breakage Tests.

a large tooth throw. The record is from an eight tooth cutter with 25 % radial immersion and 0.08 inches of axial immersion cutting 4330 steel. The step-by-step processing of the signal is shown in Figures 6-14 through 6-17. The signal, once filtered through the averaging and once per revolution filters, clearly shows the breakage of the cutter near the 30th revolution of the cutter.

The filtered and normalized signal shown in Figure 6-17 is used for the detection of the breakage. It can be seen in the figure that the breakage signal is easily seen against the remaining background noise. A threshold multiplier of five

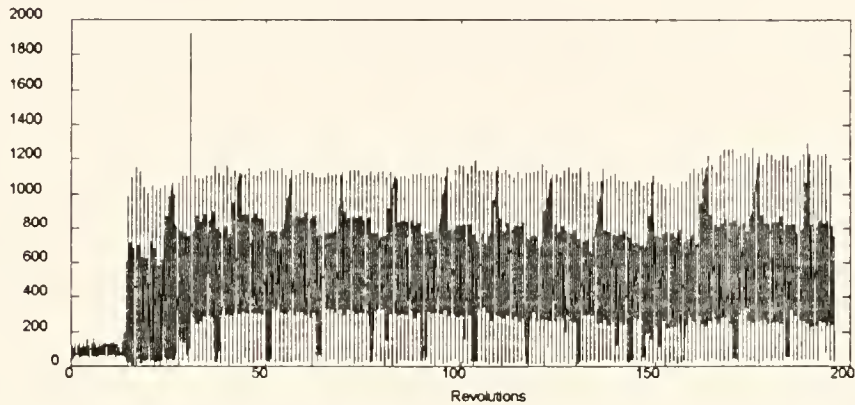


Figure 6-13 Cutting Record for in Process Cutter Breakage, 25% Radial Immersion, Eight Tooth Cutter.

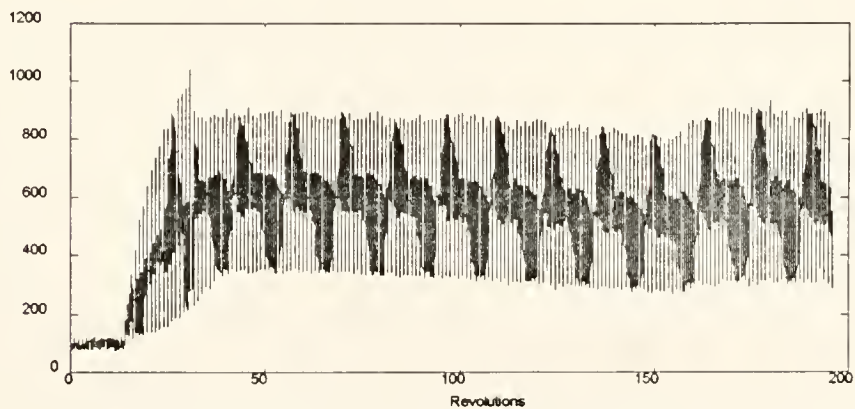


Figure 6-14 In Process Breakage Record Filtered Through the Tooth Average Filter.

was found to be sufficient to separate the breakage from the remaining noise. These threshold values could be plotted in Figure 6-17 as horizontal lines at positive and negative five. Cutter breakage is detected when the signal breaks both the positive

and negative values. An expanded view of the breakage signal for the presented record is shown in Figure 6-18. The thresholds are also plotted in the figure. The breakage signal shows a distinctive up-down pattern, with the separation of the positive and negative peaks being equal to the tooth period: fifteen samples in this case. This feature is used to allow the thresholds to be set at very narrow limits. By requiring the signal to break both the positive and negative threshold in a single tooth period, false triggers due to occasional violations of the threshold by noise can be avoided. Thus the thresholds can be narrowly set to detect lesser cutter failures such as partial chipping.

The requirement of breaking both thresholds within one tooth period proved to be necessary due to the action of the once per revolution difference filter. Because of the once per revolution differencing, any large noise spike in the data will be repeated in the opposite direction one revolution later. An example of this is shown in Figure 6-19. The figure shows a filtered record and thresholds for a 75 % radial immersion inprocess breakage test. There appears to be a violation of the thresholds near revolutions 65 and 105. Upon closer inspection, the feature near revolution 65 (expanded in Figure 6-20) is seen to be a noise spike and its negative repeated one revolution later. These strong noise spikes are caused in the present measuring system by iron chips from the cutting process passing closely to the inductance displacement probes. An expanded view of the actual breakage feature is shown in Figure 6-21. Finally, a record from an unbroken cutter is shown in Figure 6-22, showing no threshold violations.

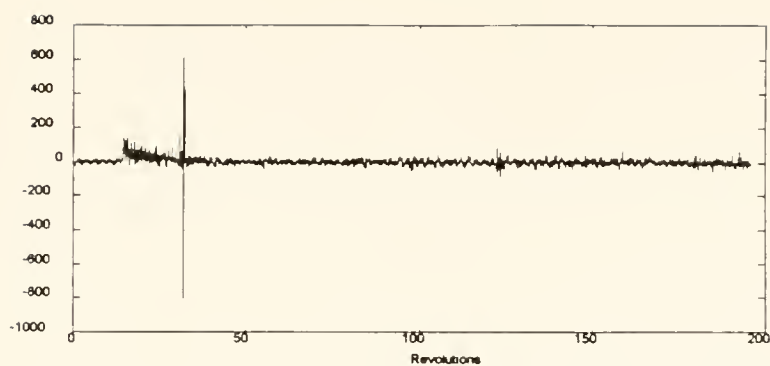


Figure 6-15 In Process Breakage Record Filtered Through the Average and Once Per Revolution Difference Filters.

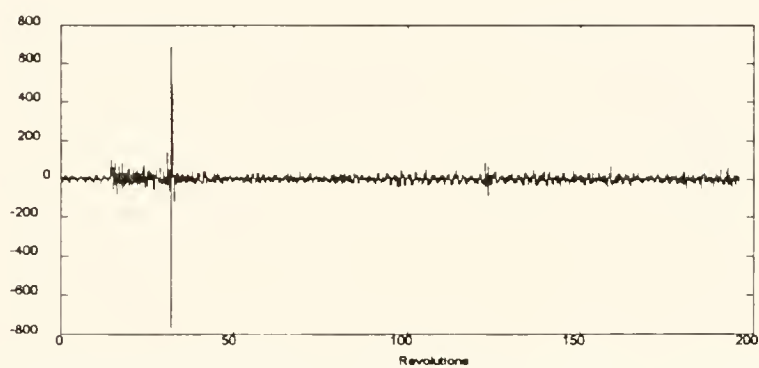


Figure 6-16 In Process Breakage Record Filtered Through the Average and Once Per Revolution Difference and DC Removal Filters.

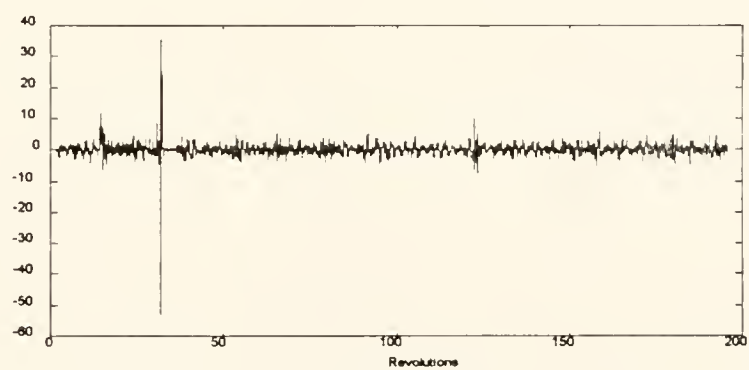


Figure 6-17 In Process Breakage Signal, Fully Filtered and Normalized to the Noise Threshold.

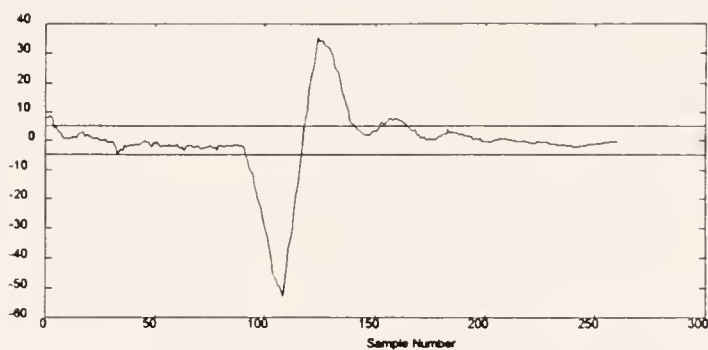


Figure 6-18 Expanded Breakage Feature and Thresholds.

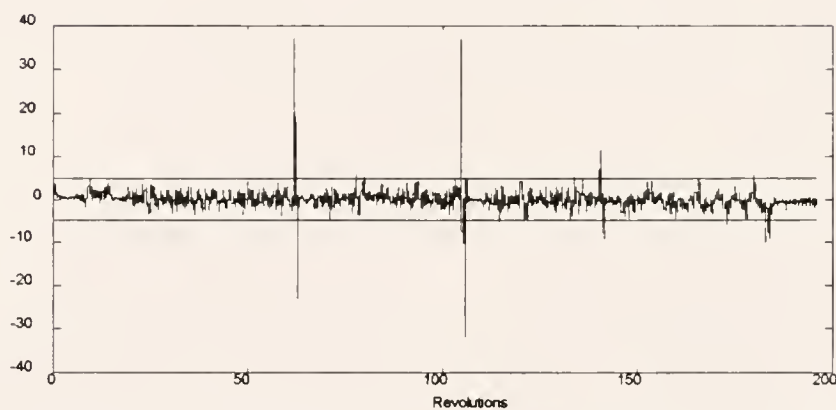


Figure 6-19 Filtered in Process Breakage Record for 75% Radial Immersion.

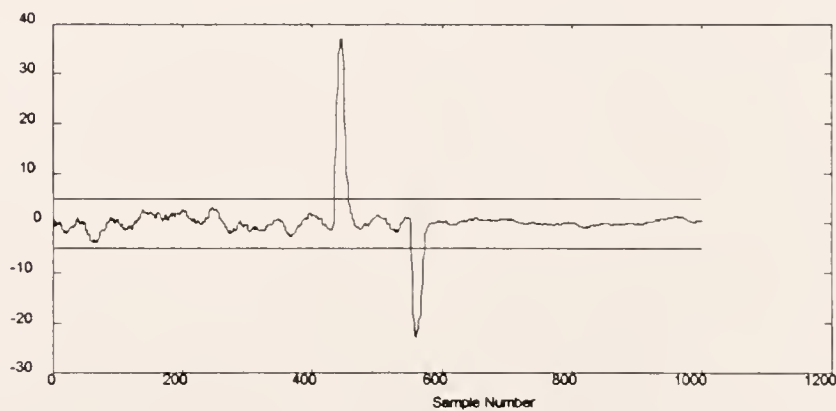


Figure 6-20 Expanded View of Threshold Violation at Revolution 65.

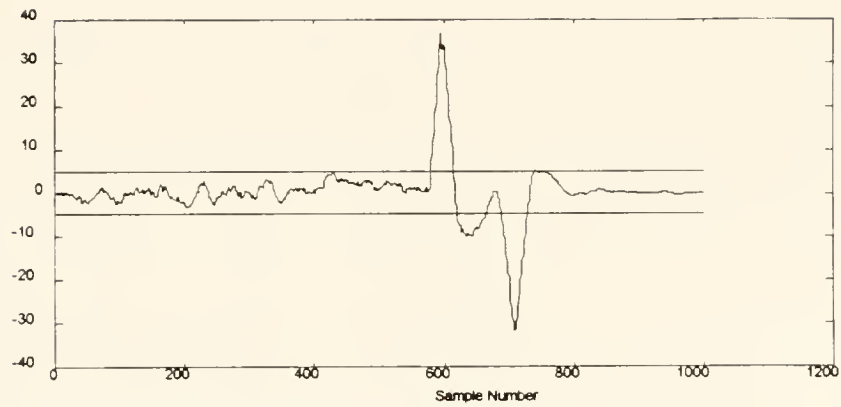


Figure 6-21 Expanded View of Breakage Feature.

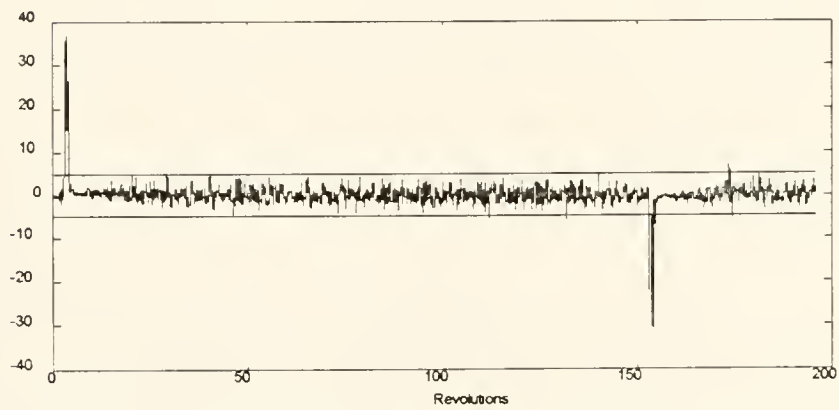


Figure 6-22 Filtered Signal and Thresholds for an Un-Broken Cutter.

CHAPTER 7

THE PROGRESSIVE DAMAGE DETECTION SUB SYSTEM

The progressive damage detection sub system is designed to detect slowly developing or previously existing cutter damage. The system is essentially a tooth throw detection system, in that it senses differences in the signal between each tooth period. It detects differences between the teeth, the system is insensitive to cutter wear, further the sub system is specifically designed to reject the signal due to cutter run-out. Because the sub system is to operate in concert with the once per revolution system, it is not required to rapidly react to sudden breakage, because, this is the task of the once per revolution sub system.

Previous detection systems have been implemented using a first difference scheme by Tarng and Tlusty [15] [19] and later Tlusty and Vierck[20]. The systems based on the first difference operate by subtracting the signal from the previous tooth period from the signal due to the present tooth period. The difference equation for this system can be written as,

$$Y_n = X_n - X_{n - \frac{120}{m}}$$

where m is the number of teeth on the cutter. The transfer function of the first difference filter is seen in Figure 7-1. The filter is a comb filter with its zeros located at the tooth frequency and its harmonics. As this is the signal content due to an

undamaged cutter and the signal due to cutter breakage is mostly located in the 'breakage region' below the tooth frequency, this is an excellent filter to separate damaged from undamaged cutters during steady state operation.

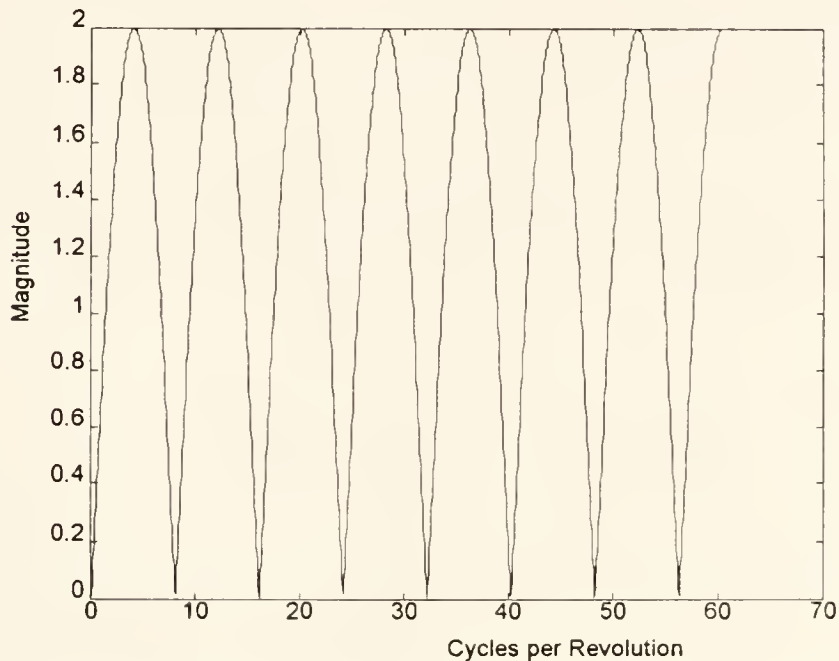


Figure 7-1 Magnitude Transfer Function for a First Difference Filter Applied to an Eight Tooth Cutter.

The filtering action of the first difference operation is shown in Figure 7-2.

The figure shows a simulated cutting operation with an eight tooth cutter first unbroken and then damaged. The entire signal is removed from the undamaged signal while the damaged cutter shows a distinct signal. This procedure can be operated in steady state cutting in a point by point fashion and completely filter the signal due to an undamaged cutter. However, during transient milling such as entry, the first difference filter passes signal content at the tooth frequency and its harmonics by the

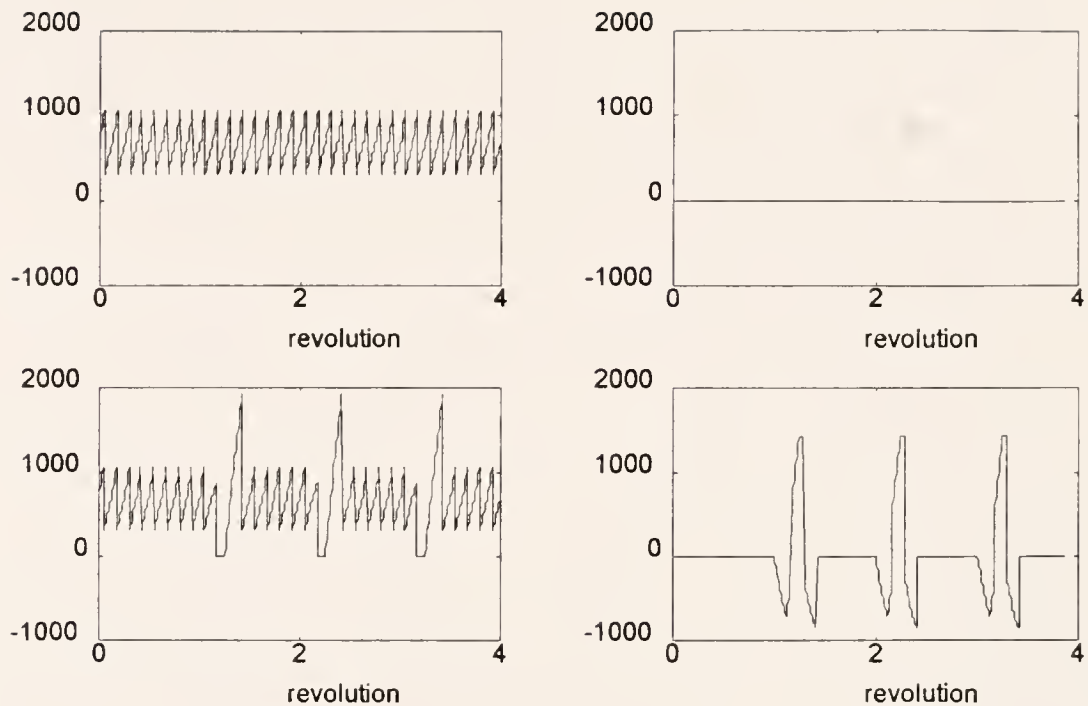


Figure 7-2 Simulated Undamaged and Damaged Cutting Signals, and the Signals Filtered Through the First Difference Filter.

same mechanism as the once per revolution difference filter. Because of this, it is convenient to first apply an averaging filter to the signal. Tarng and Vierck both used an average over the tooth window, and then an average over the next window. With the tooth by tooth average, the remaining signal form is affected by where in the signal the tooth average is placed.

The form of the out put signal from the first difference type systems has been shown to be very dependent on the radial immersion of the cutter. This radial immersion is not known during the cutting and hence, must be extracted in some way from the cutting signal. Vierck used the dc level of the signal divided by the peak-to-peak variation of the raw signal as an indication of the radial immersion, and then

varied the threshold setting based on this measure. Further, the first difference systems as implemented are strongly affected by cutter run-out.

It is desired to create a breakage detection system that is not sensitive to the radial immersion of the cutter or to cutter run-out. To do this, the signal from the cutting process is decomposed. The signal is made up of the superposition the signal of an ideal cutter with all equal teeth and the signal content due to tooth throw and run-out. The content of the ideal cutter is located in the per revolution domain at dc, the tooth frequency and its harmonics. The content due to the ideal cutter can be filtered from the signal by the application of a comb filter that only passes these frequencies. Such a filter can be written as follows:

$$y(n) = \frac{1}{m} [x(n) - x(n-120) + y(n-\frac{120}{m})]$$

This filter has zeros at the dc and the run-out frequency and its harmonics, due to the 120 delay differencing of the input signal. By adding back in the output, y, with a delay equal to the tooth period, poles are placed at dc and the tooth frequency and its harmonics. In this way, the poles cancel the zeros at dc and the tooth frequency and its harmonics. This is the same form as the 'tooth force composition filter' as presented by Walters [9]. Walters used the filter to compose the cutting force signal for an ideal cutter from the force of a single tooth. The transfer function of this filter is shown in Figure 7-3. It can be seen that the filter has zeros at the run-out frequency and its harmonics, thus any signal due to tooth throw or run-out will not be passed by the filter and the output will be of the ideal cutter.

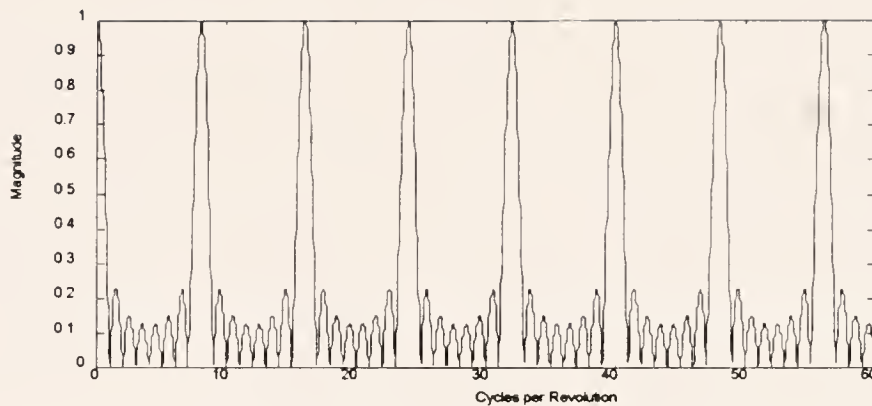


Figure 7-3 Magnitude Transfer Function of Comb Filter to Extract Ideal Eight Tooth Cutter Signal.

The action of this comb filter is demonstrated on simulated cutting force signals. Figure 7-4 shows a simulated cutting force signal of an eight tooth cutter, cutting with 25% radial immersion. The cutter is simulated to have 25% tooth throw and 25% run-out. The per revolution frequency content of the signal is shown in Figure 7-5. This shows the content at the run-out and its harmonics due to the cutter run-out and throw, and the content at dc and the tooth harmonics due to the ideal part of the cutter.

The simulated signal is filtered through the comb filter to extract the component due to the ideal cutter. The out put of this process is shown in Figure 7-6A. For comparison, the simulated cutting force signal of an ideal cutter is shown as Figure 7-6 B. The two signals are shown on a slightly expanded scale to reveal their form. The signals appear to be identical, showing that the component of the cutting force due to the ideal cutter is extractable from the cutting force signal.

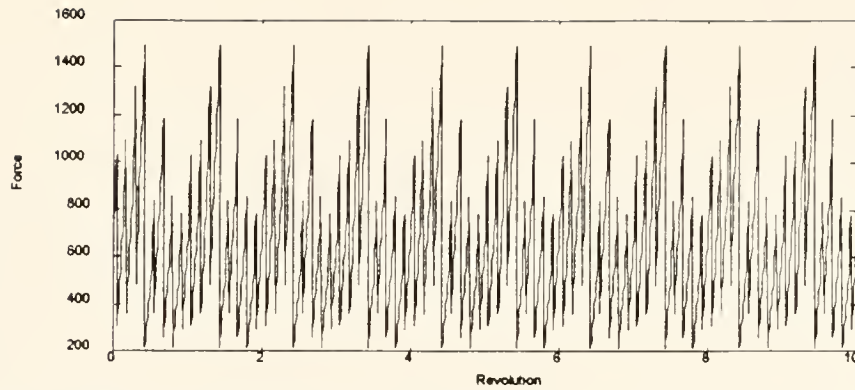


Figure 7-4 Simulated Cutting Force Signal, Eight Tooth Cutter 25% Radial Immersion, 25% Tooth Throw, 25% Run-out.

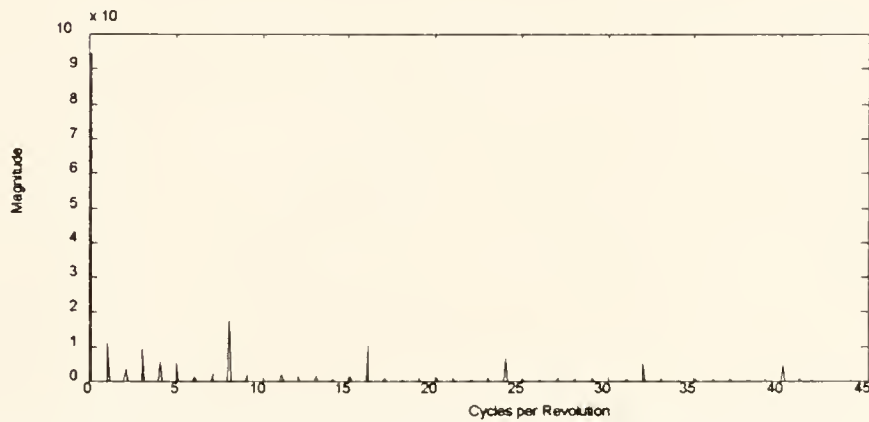


Figure 7-5 Spectrum of Simulated Force Signal, Eight Tooth Cutter, 25% Radial Immersion, 25% Tooth Throw, 25% Run-out.

The difference between the ideal cutter signal and the actual cutter signal is due to tooth throw and cutter run-out and whatever noise is present in the measured signal. The difference of the extracted ideal cutter signal and the unfiltered simulated signal is shown in Figure 7-7. The spectrum of this signal is seen in Figure 7-8. The spectrum is seen to have content only at the run-out and its harmonics not including the tooth frequency and its harmonics.

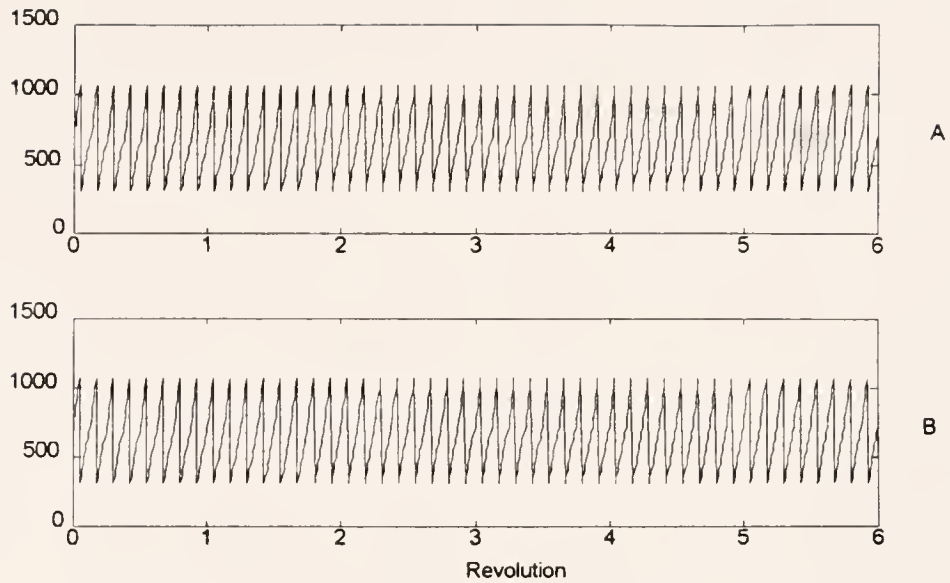


Figure 7-6 Simulated Cutting Force Signals for Eight Tooth Cutter, A) Extracted Ideal Cutter Signal From Comb Filtering Cutter With 25% Tooth Throw and 25% Run-out Signal B) Simulated Ideal Signal.

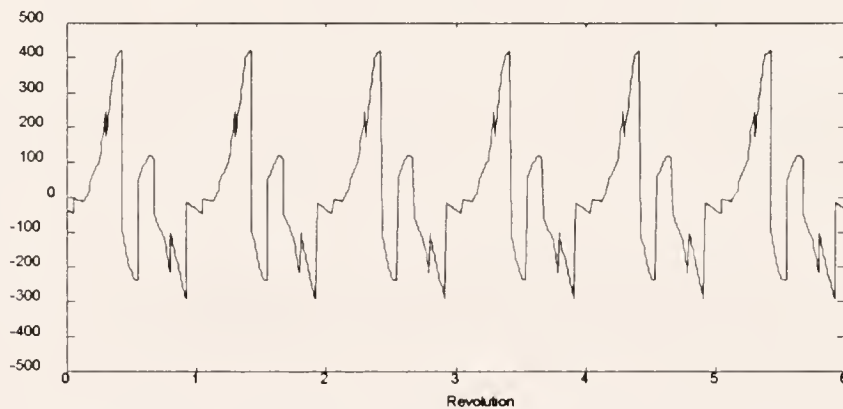


Figure 7-7 Difference Signal of the Extracted Ideal Cutter and Simulated Force Signal.

Finally, to remove the component due to cutter run-out the remaining signal is filtered through a high pass FIR filter with the cut off frequency slightly higher than the once per revolution signal. A high order FIR filter is used for this purpose. The

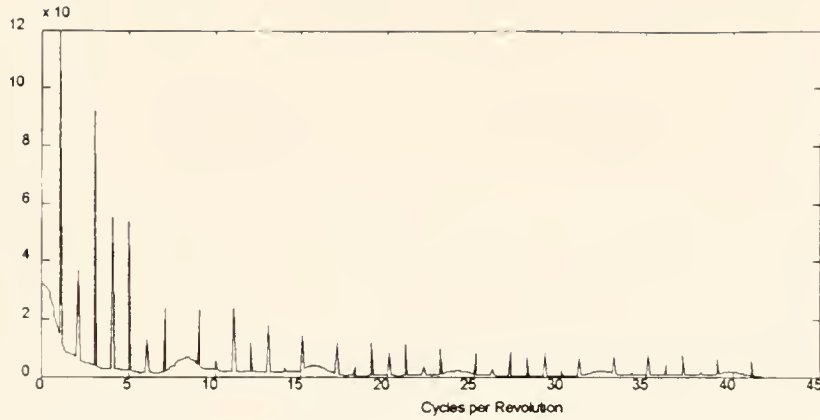


Figure 7-8 Spectrum of Difference Signal of the Extracted Ideal Cutter and Simulated Force Signal.

high order is needed with a FIR filter to get a sharp cutoff, thereby avoiding attenuating other parts of the breakage region too severely. An FIR filter is used to avoid the phase distortion that would be introduced by an infinite impulse response filter, which would distort the form of the breakage signal. The form of the FIR filter can be written as follows,

$$Y_n = \sum_{k=0}^L b_k X_{n-k}$$

where b_k represents the coefficients of the filter, and L is the filter order. Because the signal is synchronously sampled and of course the run-out frequency is always once per revolution, the cutoff frequency of the designed filter is constant for any speed and number of cutter teeth. The digital cutoff frequency is the desired cutoff frequency divided by the Nyquist frequency. Thus, for the run-out, the digital cut off frequency is $1/60$ (0.01666). To place the cut off just above the run-out, assuring that it is thoroughly removed, the cut off frequency used is 0.025, where 1.0 represents half the sampling frequency. The magnitude transfer function plot for a 240 order high pass

filter is shown in Figure 7-9. The phase transfer function is linear for any FIR filter, representing a simple time delay.

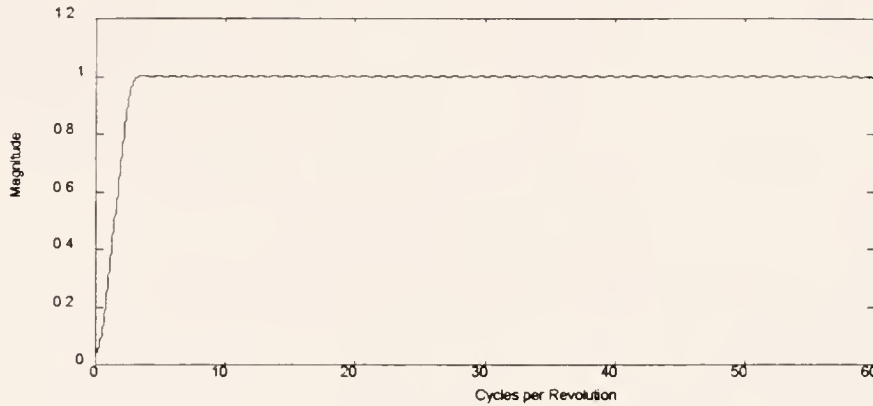


Figure 7-9 Magnitude Transfer Function of a 240 Order High Pass Filter with Cut Off Frequency of 0.025, where $f_s / 2 = 1$.

The signal is also filtered by a running average filter with the window length equal to the tooth period. The averaging filter is included to remove signal content at the tooth frequency and its harmonics during transient milling conditions, as was done in the sudden breakage detection subsystem. The difference equation of the running average filter can be written,

$$Y_n = \frac{m}{120} * \sum_{i=0}^{\frac{120}{m}-1} X_{n-i}$$

The transfer function of this filter is seen in Figure 7-10.

A single filter that performs the entire filtering function of averaging and high passing the signal can be formed by convolving the coefficients of the two filters with each other. The overall transfer function of this single filter is shown in Figure 7-11.

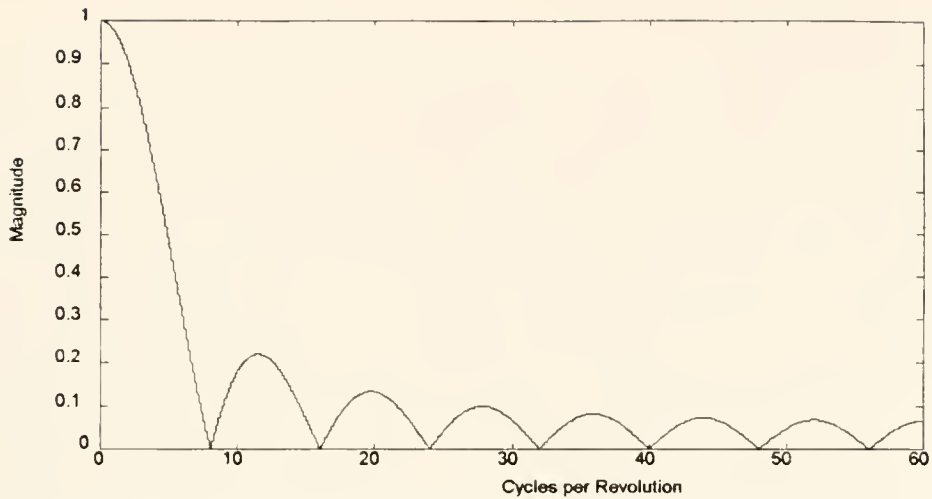


Figure 7-10 Magnitude Transfer Function for a Running Average Filter, with Window Length Equal to the Tooth Period for an Eight Tooth Cutter.

The filter is seen to remove any remaining dc component, the run-out, and the tooth frequency and its harmonics. The breakage region, between the run-out and the tooth frequency, is emphasized by the combined filter, while the region between the higher tooth harmonics is attenuated.

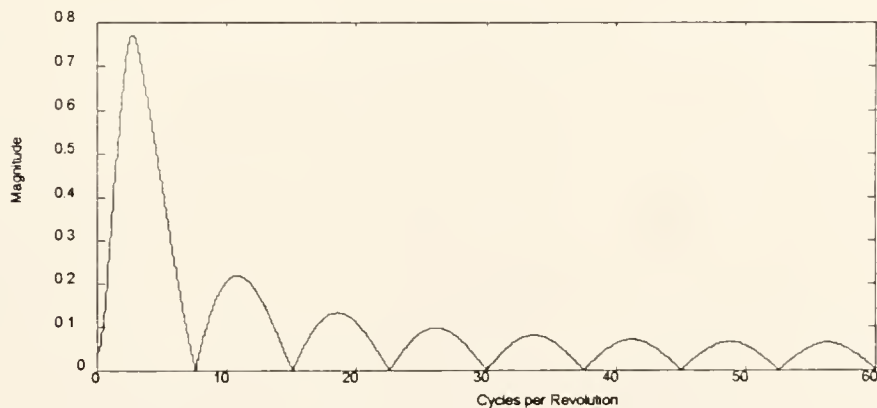


Figure 7-11 Magnitude Transfer Function of the Combined High Pass and Running Average Filter.

One shortcoming of the combined high pass averaging filter will be its application to two tooth cutters. With a two tooth cutter, it is impossible to distinguish tooth throw from run-out of the cutter. The high pass filter will remove essentially all of the breakage region of a two tooth cutter. This breakage region consists only of the run-out frequency. Additionally, there is content in the broken cutter signal at the higher run-out frequencies that is not at the tooth frequency or its harmonics. This portion of the signal is strongly attenuated by the averaging filter. Thus, the sub system may not be applicable to the case of a two tooth cutter.

The difference signal from the previously described simulation case is filtered through this combined filter and displayed as Figure 7-12. The random tooth throw is evident in the figure.

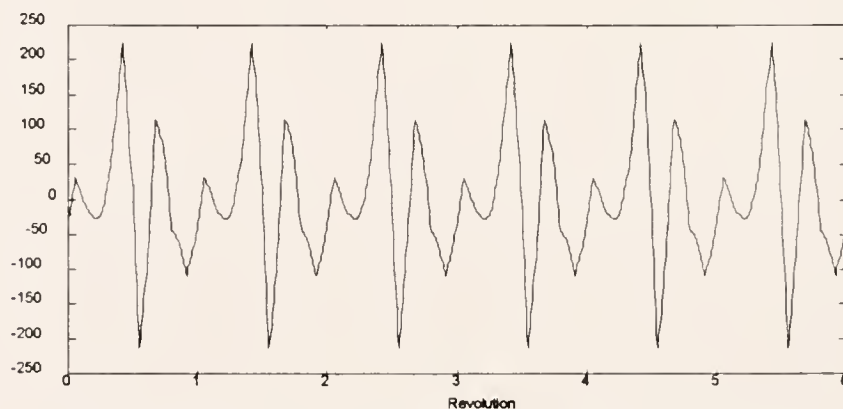


Figure 7-12 Fully Filtered Force Simulation Data, Eight Tooth Cutter, 25% Radial Immersion, 25% Tooth Throw, 25% Run-out.

The effect of the filtering procedure on simulated broken cutters with varying radial immersions is shown in Figure 7-13 and 7-14. Figure 7-13 shows the unfiltered force signals while the filtered signals are shown in Figure 7-14. All signals are from

simulated cuts with an eight tooth mill. The simulations are of 25, 50 and 100% radial immersion, and the cutters are simulated to have 50% run-out to demonstrate the system's ability to reject the cutter run-out from the signal.

In Figure 7-14, it can be seen that the form of the cutter breakage signal is affected by the radial immersion of the cutter. Specifically, the breakage signal takes on a more rounded form as the radial immersion is increased. The overall magnitude of the breakage signal does not vary greatly with radial immersion. The remaining signal is the force variation due to the tooth throw or breakage. The overall magnitude is therefore directly dependent on the tooth forces. The tooth force can be expressed by the following equation:

$$F_t = k_s * a * b * \sin\theta$$

where k_s is the cutting stiffness of the work piece material, a is the axial immersion of the cut and b is the feed-per-tooth. The value of $(k_s * a * b)$ acts as a scaling factor on the cutter breakage signal. By dividing by this quantity, the breakage signal may be normalized so that the signal represents the percentage of tooth throw present on the cutter.

The value of $(k_s * a * b)$, now referred to as the scaling factor, is not directly available from the cutting signal. Further, it cannot be calculated, as it may contain a calibration factor between the actual force and that measured during the cutting process. Therefore, it is recommended to preform a calibration cut to determine the

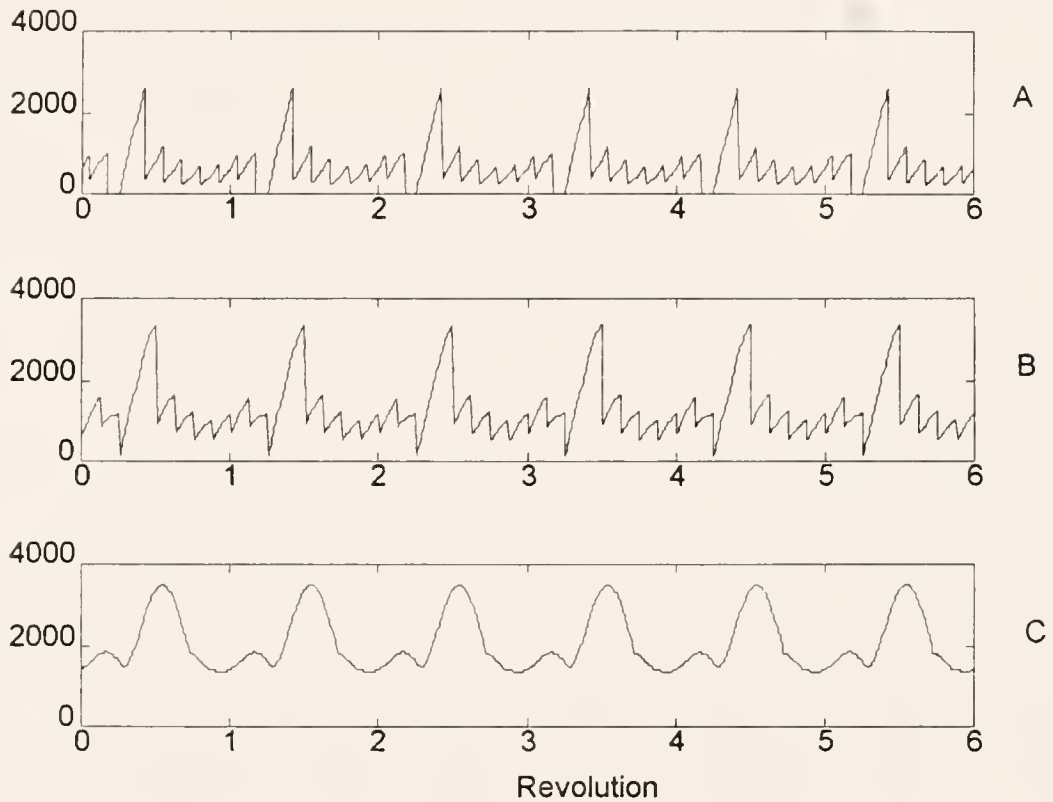


Figure 7-13 Simulated Cutting Force Signals for an Eight Tooth Cutter with a Broken Tooth and 50% Run-out. A) 25% B) 50% C) 100% Radial Immersion.

the value of the scaling factor. The scaling factor will change linearly with the chip load and the axial immersion.

The calibration cut is preformed preferably at the same axial depth and chip load as the cutting process, with a known unbroken cutter. The cut is performed at a known radial immersion in order to allow the extraction of the scaling factor from the cutting data. By knowing the radial immersion of the cutter and the number of cutter

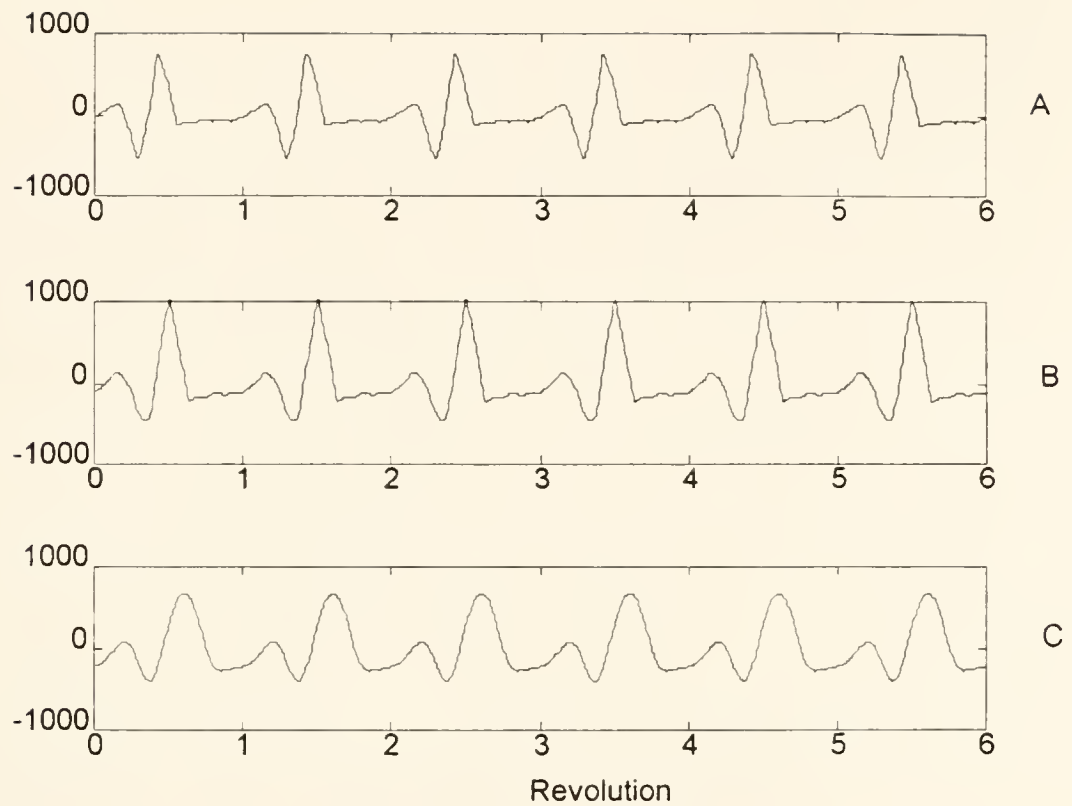


Figure 7-14 Filtered Simulated Cutting Force Signals for Eight Tooth Cutter with a Broken Tooth A) 25% B) 50% C) 100% Radial Immersion.

teeth, the form of the cutting force signal is known. Therefore, the scaling factor can be extracted from the data. If a 50% radial immersion cut is used, the scaling factor can be related to the peak-to-peak force pattern of the calibration cut by the following correction factors. These factors are determined by calculating the difference of the vector sum of all the tooth forces at the location of maximum force and the location of

minimum force for the cutter to determine the relation between the peak-to-peak force variation and the scaling factor.

Calibration Correction Factors for 50% Radial immersion.

Number of Teeth.	Correction Factor.
2	1
3	1
4	1
6	1.215
8	1.144

The procedure to measure the calibration factor is then to make the calibration cut at the 50% radial immersion. The steady state force data is filtered by the comb filter to extract the signal due to an ideal cutter. The peak-to-peak variation of this signal is determined and this value is multiplied by the correction factor based on the number of teeth on the cutter to obtain the scaling factor. The out put of the filtering system during cutting is then divided by the scaling factor to normalize the signal to the tooth force.

Simulated cutting force data for a 50% radial immersion calibration cut is shown in Figure 7-15. The data has 5% tooth throw and run-out to simulate a signal that is actually attainable. The data is for a simulated cut with the same axial immersion and chip load as the previously shown 25% radial immersion data, thus the

scaling factor from this data is directly applicable to the previous data. The filtered signal is shown in Figure 7-16, where the peak-to-peak variation is seen to be equal to 820 with the scaling factor being 938.

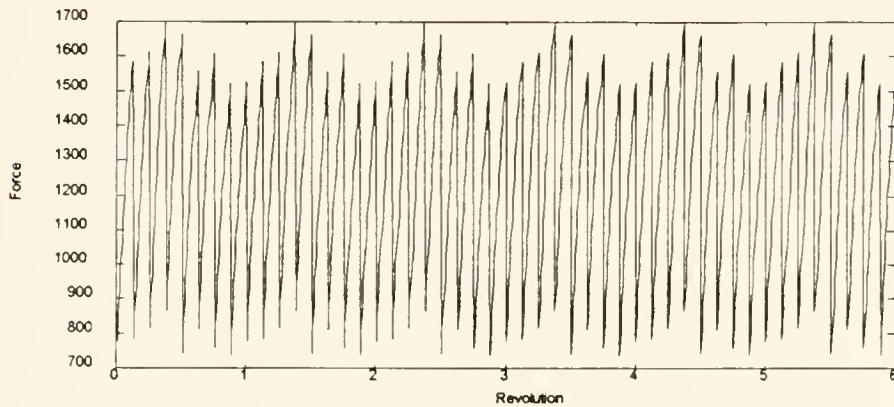


Figure 7-15 Simulated Calibration Signal, 50% Radial Immersion, Eight Tooth Cutter.

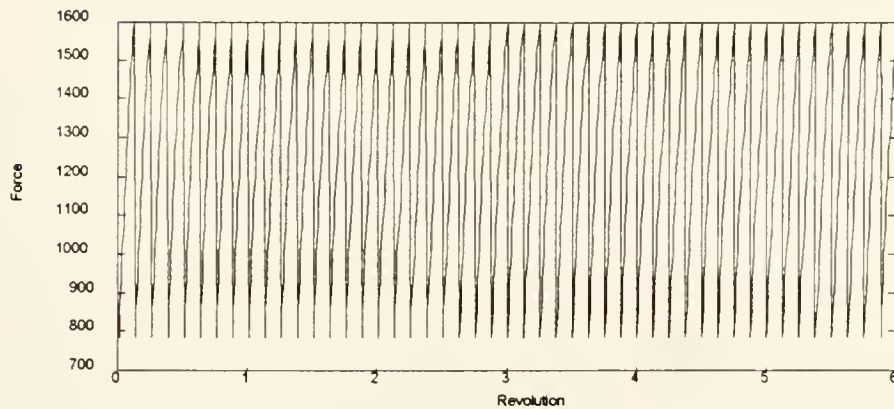


Figure 7-16 Filtered Simulated Calibration Signal, 50% Radial Immersion.

The fully filtered signal from the previously shown simulated cut is shown normalized by the calibration factor in Figure 7-17. Originally, the cutter was

simulated to have 25 % tooth throw. The filtered and normalized signal shows a tooth throw variation of just less than 25% . The slight attenuation of the variation is due to the passing of the signal through the high pass and averaging filters which slightly attenuates the breakage region.

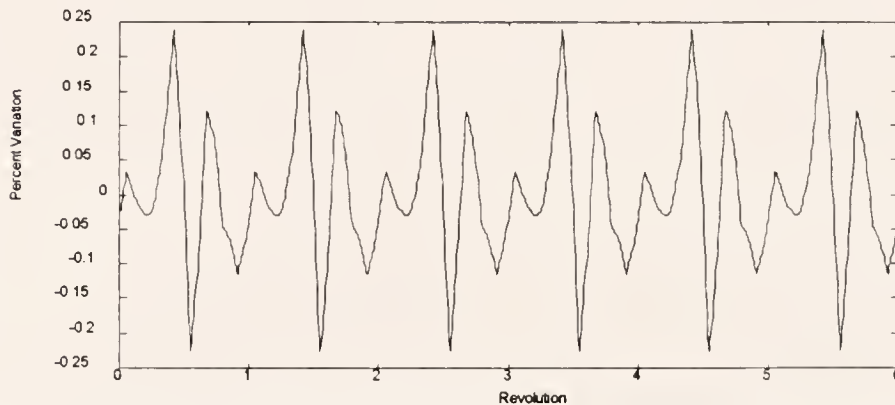


Figure 7-17 Filtered and Normalized Signal from Simulated 25% Radial Immersion Cut, Eight Tooth Cutter, 25% Tooth Throw.

The performance of the progressive damage sub system is demonstrated on actual cutting data. The cutting data presented is from an eight tooth face mill cutting cast iron. The data was collected from inductance displacement sensors measuring the displacement of the spindle in two orthogonal directions on a Sundstrand series twenty Omnimil. This displacement signal is taken as representative of the force signal from the cutting process. The signal was collected in synchronization with the spindle rotation with 120 samples per revolution. In all cases, the signal shown is the vector sum of the two displacement signals.

A 50% radial immersion up milling cut is used as a calibration cut for the collected data. The unfiltered data for this cut is shown in Figure 7-18. The break in

the middle of the data record is due to two sections of data being recorded. The entry and initial steady state cutting portion were recorded, and then, during the same cut, the exit was recorded. The break is a section during the starting and stopping of the recording.

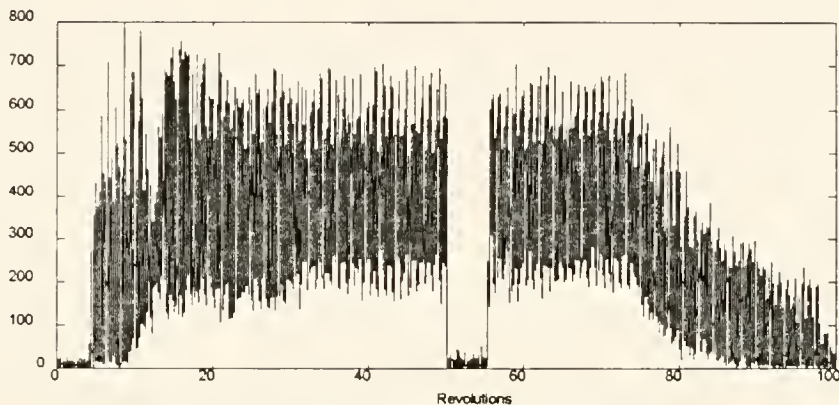


Figure 7-18 Cutting Force Record for 50% Radial Immersion Cut. Eight Tooth Cutter.

A short section of the data from the steady state portion of the cut, around the 40th revolution, is shown in Figure 7-19 after being passed through the comb filter. The peak-to-peak variation is seen to be 300. The units are not known, as the displacement versus force calibration is not known, and the data was recorded only in terms of digital bits of the range of the data acquisition. The units are unimportant as they will be divided out in the normalization of the signal to give only a percent variation. The scaling factor is therefore equal to 343.2.

The cutting force signal from a 25% radial immersion case with an unbroken cutter is shown in Figure 7-20. The filtered and normalized signal from this cut is

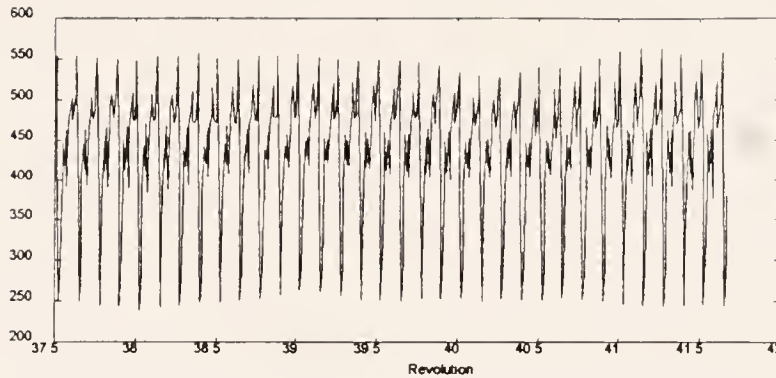


Figure 7-19 Filtered 50% Radial Immersion Calibration Cut Signal.

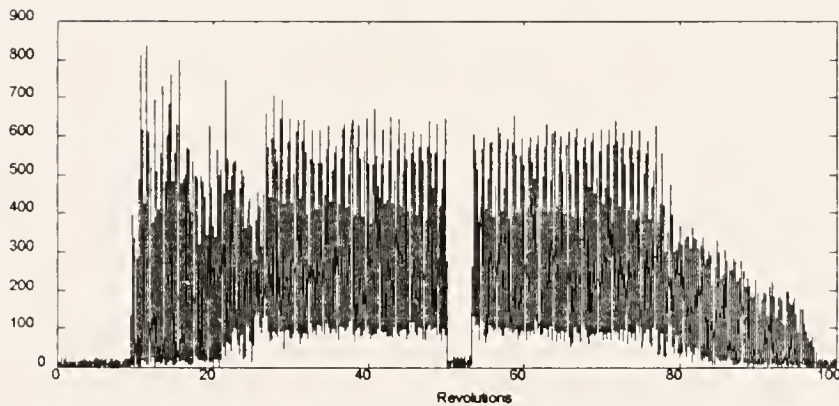


Figure 7-20 Unfiltered Cutting Force Signal for Unbroken Eight Tooth Cutter, 25% Radial Immersion.

shown as Figure 7-21. The cut was performed at 600 RPM with a feed rate of 50 inches per minute, resulting in a chip load of 0.010 inches per tooth. The unbroken cutter was set with a tooth throw tolerance of 0.001 inches. Therefore, the tooth throw allowed on the unbroken cutter is 10%.

The filtered and normalized cutting force data for a 25% radial immersion case with one insert removed to simulate cutter breakage is shown in Figure 7-22. Note the scale is increased from the previous figure.

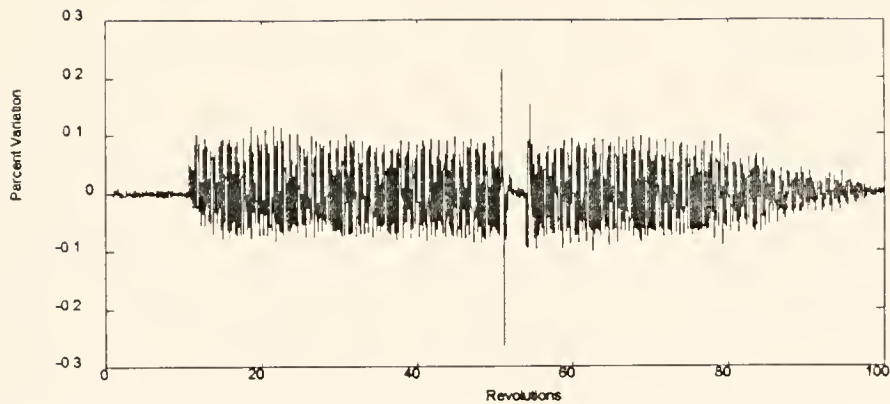


Figure 7-21 Filtered and Normalized Signal for 25% Radial Immersion Unbroken Eight Tooth Cutter.

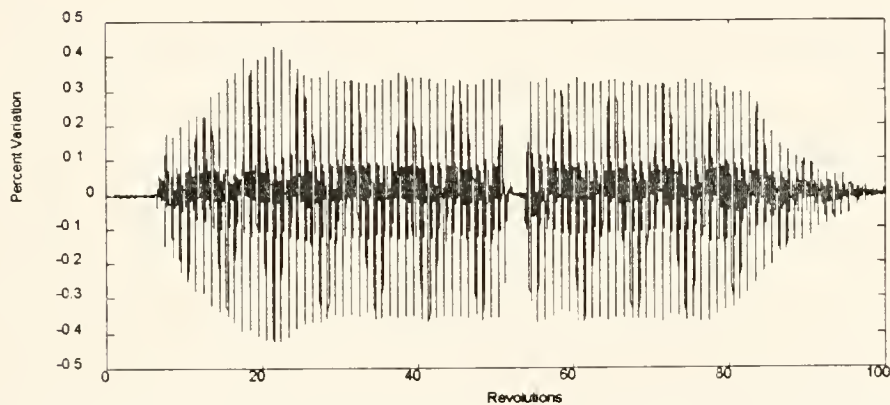


Figure 7-22 Filtered and Normalized Cutting Force Signal for Broken Eight Tooth Cutter, 25% Radial Immersion.

Fully filtered and normalized output signals for the eight tooth cutter for 50, 80 and 100% radial immersion are shown in Figure 7-23. All three plots are for unbroken cutters with tooth throw allowance of 10%. The corresponding filtered and normalized signals for broken cutters under the same cutting conditions are shown in Figure 7-24. The plots for the unbroken cutters all show approximately 15% variation, slightly higher than expected from the allowable tooth throw. The broken versus unbroken cutters are distinguishable by setting thresholds at plus and minus 20% tooth throw. A

strong difference is seen in the transient milling sections of entry and exit, indicating that there is still some dependence on the radial immersion of the cutter. The high radial immersions with multiple teeth involved in the cut tend to average the tooth throw over multiple teeth. This makes the detection of breakage in the deep radial immersion much more difficult than in the lower. The method does not exhibit any reduced performance during the exit phase of deep radial immersion cuts. These cases produced reduced performance in the first difference method of Vierck due to the 'exit honk' during the interrupted cut produced in the deep radial immersion exit.

By requiring the signal to violate both the positive and negative thresholds on multiple revolutions, the thresholds may be set very low. This allows detection of smaller changes in the cutters condition, while protecting against false triggers from occasional noise.

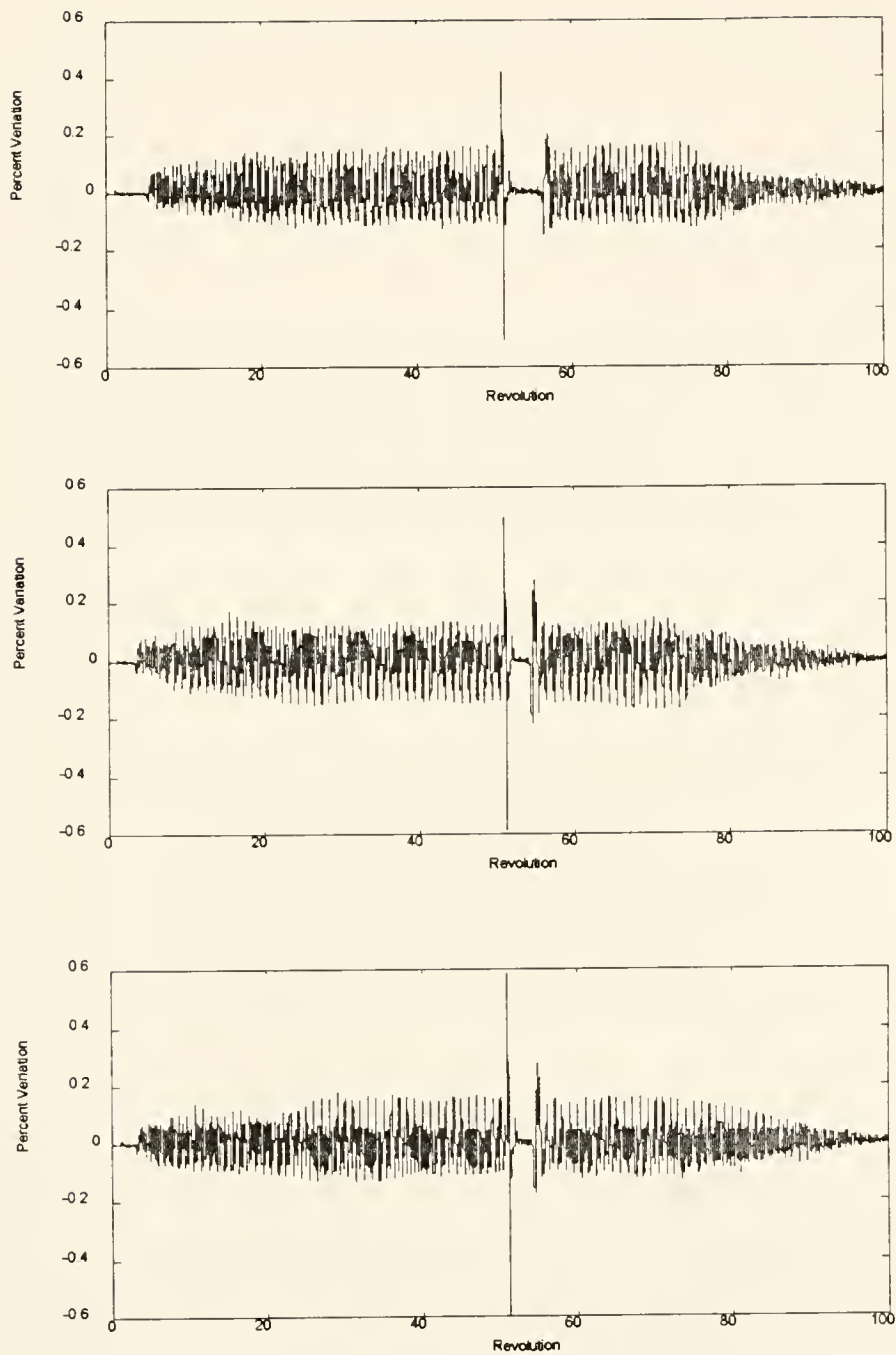


Figure 7-23 Filtered and Normalized Signals for Unbroken Cutter, Radial Immersions
A) 50% B) 80% C) 100%.

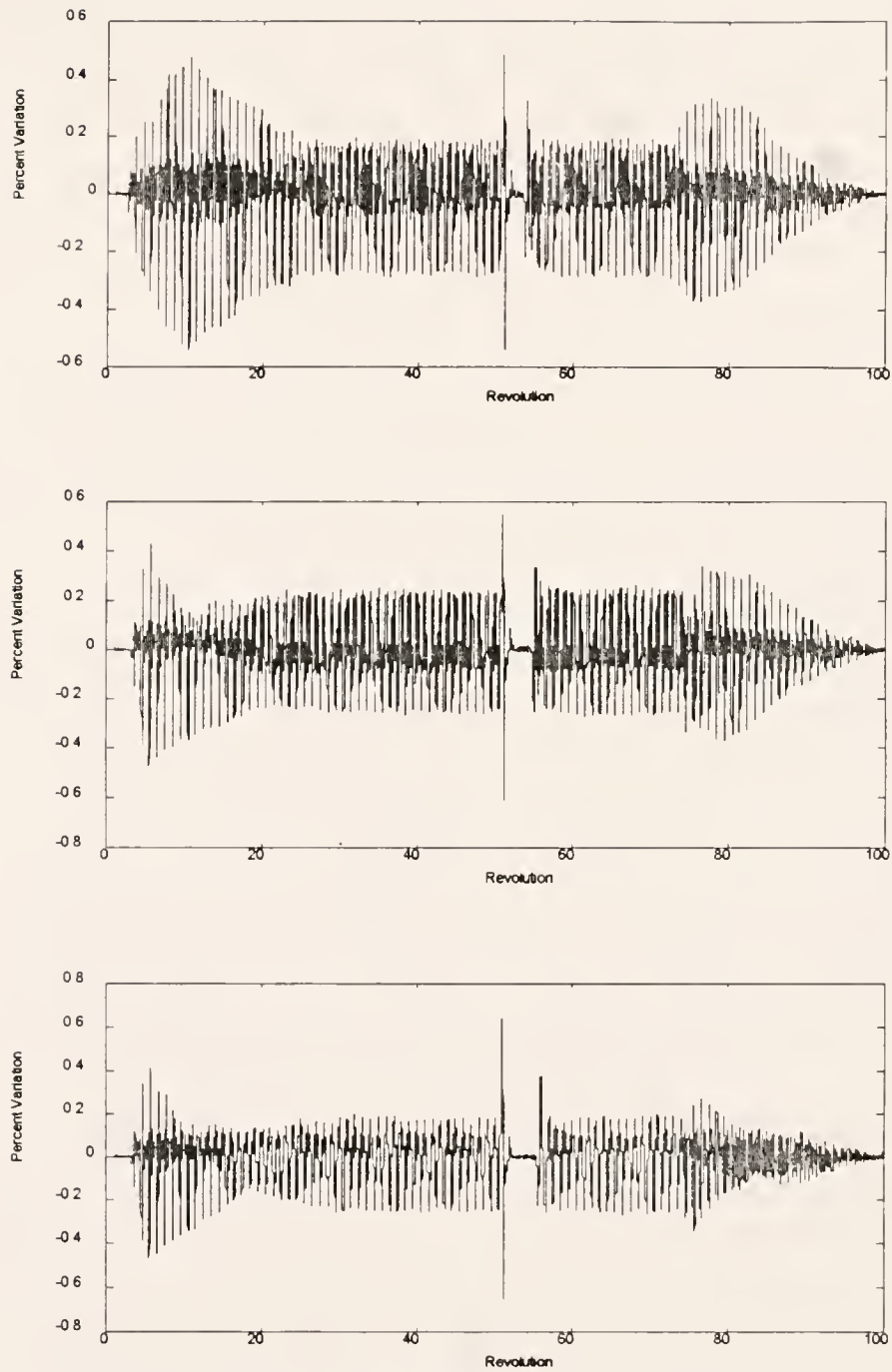


Figure 7-24 Filtered and Normalized Signals for Broken Eight Tooth Cutter. Radial Immersion A) 50% B) 80% C) 100%.

Similar results are presented for a six flute, two inch diameter end mill cutting aluminum. Figure 7-25 shows the filtered and normalized results for an unbroken end mill, cutting with 20% radial immersion. The end mill was used with a chip load of 0.004 inches. The signal from the unbroken end mill indicates a tooth throw of 10%. The corresponding data for a similar end mill with a broken flute are shown in Figure 7-26.

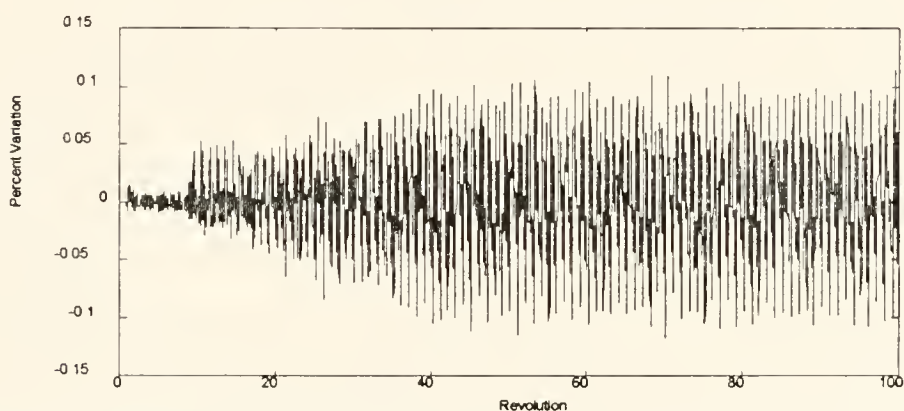


Figure 7-25 Filtered and Normalized Signal for an Unbroken Six Flute End Mill in Aluminum, 20% Radial Immersion.

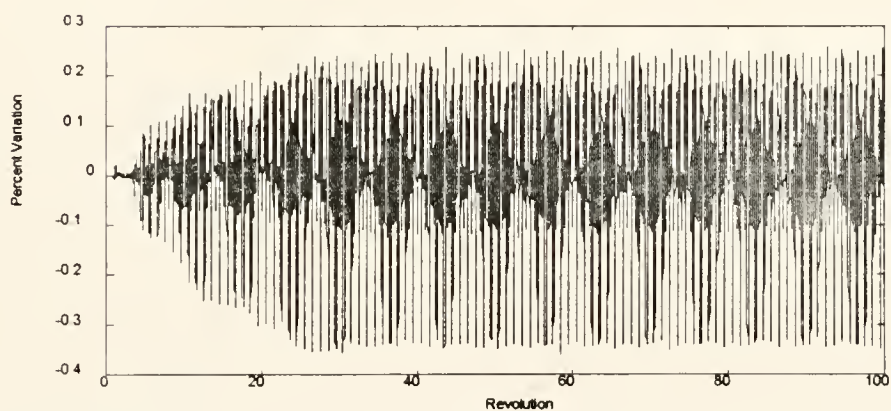


Figure 7-26 Filtered and Normalized Signal for Broken Six Flute End Mill in Aluminum, 20% Radial Immersion.

CHAPTER 8

CONCLUSIONS AND FUTURE WORK

The primary focus of this work was the illustration of the application of digital filtering techniques to the unmanned supervision of milling. For these techniques to be applied effectively, the structure of the milling signal must be fully understood. To aid in this understanding, the particular features of the cutting force signal were examined, including the effects of cutter run out and tooth throw on the milling force spectrum. The spectral content due to these features was demonstrated and utilized in the formation of filtering schemes to aid in the detection of milling chatter and the detection of milling cutter breakage.

The effectiveness of the filtering techniques applied was shown to be very dependent on the method of data acquisition from the milling process. The signal used for the detection of chatter is acquired in a time synchronous fashion, eg. at a constant sampling frequency. The consequence of utilizing a time domain signal, while attempting to filter from it signals that are synchronous with the spindle rotation, is that of the misplacement of the filter notches and hence reduced filter performance. The signal used for the detection of cutter breakage is sampled in a rotation synchronized fashion, leading to the ability to design filters in the per revolution domain as opposed to the frequency domain. This provides filters to have ideal placement relative to rotation synchronous signal components which comprise the

majority of the milling force signal, leading to simplification and improved performance of the filters.

Filters for Chatter Detection

The performance of the revolution referenced comb filter was investigated in its use in filtering a sound signal utilized in the detection of milling chatter. The effects of notch misplacement due to the use of a non-synchronous signal were demonstrated and quantified. The filter performance can be determined from the amount of misplacement of the notches. This misplacement is based on the ratio of the sampling frequency to the frequency of the run out harmonics to be filtered. Therefore, for a desired filter performance up to a predetermined frequency, the required sampling frequency of the system can be directly determined.

The performance of the revolution referenced comb filter was investigated in the case of spindle speed variation as would be encountered in entering a cut. The filter notch misplacement is aggravated during these transients. The notch misplacement is most severe at the high frequencies. The loss of filter performance is balanced by a natural blurring of the high frequency components in the frequency domain during these transient conditions. These two effects counter each other, finally revealing good filter performance during the transient.

In an attempt to improve the revolution referenced comb filter, a series of poles was added to narrow the filter notches, creating an infinite impulse comb filter. This was done to avoid filtering of a valid chatter signal that was close to a run out

harmonic and therefore a filter notch. The narrowing of the filter notches was shown to improve the filter performance in these cases when the filter had no notch misplacement. The IIR comb, with its narrow notches, was shown to be ineffective in the existence of notch misplacement.

A two path filtering scheme was outlined for the use of two comb filters, one filtering the run out harmonics in the low frequency region, and one filtering only the tooth harmonics at higher frequencies. To separate which harmonics were to be removed, the distribution of the run out harmonics in the milling force spectrum versus the radial immersion was evaluated. To enact the tooth frequency comb, it was shown that the filter notch misplacement would be increased by fifty percent as compared to the run out comb. This would require that the supervision system have a fifty percent higher sampling frequency to achieve the same performance as the run out filter, for the same frequency range of interest. Further, the radial immersion of the cutter would be required in setting the transition point between the two paths.

The major obstacle in improving the performance of the revolution referenced comb is that of the notch misplacement, emanating from the time domain sampling of the data. This notch misplacement causes the IIR comb to be ineffective and requires an increased sampling frequency for the two path filtering method.

Future work to improve this could be done in three ways: The sampling frequency of the present system may be increased to reduce the notch misplacement. A method of interpolating data points could be used so that integer numbers of delays were not required of the comb filters. This may be more computationally expensive

than is practical for a real time monitoring system. Finally, the system could be changed to synchronous sampling. This eliminates the detriment of notch misplacement and would allow either of the presented filtering systems to be used without the problems associated with the notch misplacement. As the problem of filtering a valid chatter signal with the revolution referenced comb is relatively rare, and is speed dependent, the present procedure of varying the speed slightly based on the level of an unfiltered signal may be the most practical solution.

Filters for Cutter Breakage Detection

Filtering techniques were applied to the problem of in process detection of milling cutter breakage. The signals used in the cutter breakage routines are synchronously sampled with the revolution of the spindle leading to improved filter performance compared to non-synchronous sampling. The synchronously sampled signal was shown to be transformed into the per revolution domain by the fourier transform. This allows filters to be designed with constant properties regardless of spindle speed. The detection of cutter breakage was split into two parts; the detection of catastrophic insert failure during milling and the detection of slowly developing or previous cutter damage.

The subsystem designed to detect catastrophic insert failure is based on a comb filter removing all of the cutting force signal. The signal from the breakage is passed only at the point of breakage and then filtered from the remaining signal. This system also includes averaging filters to allow the system to operate properly during transient

milling stages. The system is based completely on filtering as opposed to algorithmic techniques to allow efficient implementation on digital signal processing hardware.

Due to the difficulty of producing controlled insert breakage, the system was tested only on data with complete insert failure during heavily loaded cutting conditions. Future work on the sub system should investigate its sensitivity to lesser breakages. This will require the development of techniques for introducing a more controlled breakage during the cutting procedure.

The sub system designed to detect progressive or pre-existing cutter damage is also based on filtering that is efficiently processed by digital signal processing hardware. The system is based on extracting and removing the portion of the signal due to an ideal cutter, thus leaving only that content due to tooth throw or cutter breakage. Filters are also included to handle transient milling cases and make the system insensitive to cutter run out. The subsystem relies on a calibration cut to normalize the resulting filtered signal during the actual cutting operation. The sub system was shown to be able to extract the signal due to tooth throw and cutter breakage. The resulting difference between good and broken cutters was most obvious during low immersion and transient milling conditions.

The signal was not independent of radial immersion, which is desired. Future work should concentrate on normalizing the signal with the extracted radial immersion. Radial immersion may be inferred from the height of the tooth frequency component versus that of the steady state portion of the signal. These signals should be extracted from the ideal cutter signal as filtered from the on-line data. This is similar to the

approach of Mr. Vierck. Perhaps the two methods could be combined to provide improved performance.

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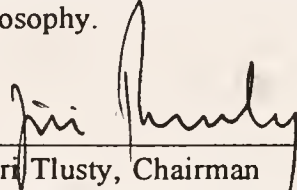
BIOGRAPHICAL SKETCH

The author was born, at a very young age, on June 11, 1963. After a period of bumbling and stumbling aimlessly through childhood development, the author entered the University of Florida. During his tenure at the university the author met the lovely Catherine M. Russo. He earned his bachelor's degree with high honors in mechanical engineering in May of 1986. He left the university for a job at Pratt Whitney Aircraft, vowing never to return.

The author began his masters program at the University in August of 1987. The lovely Catherine had moved to Manhattan, New York. While living at the Purple Porpoise, the author enriched the shareholders of AT&T and the airline industry pursuing Catherine, and enriched Anheuser Busch pursuing other interests. He managed to complete his Masters of Science in August of 1989. He left the University for Houston, Texas, vowing never to return.

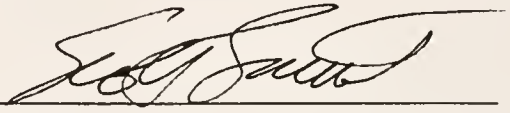
The author began a doctorate program at the University in August 1991; AT&T and the airline industry rejoiced. The lovely Catherine returned in 1993 for their marriage, bringing with her their beautiful daughter Anastasia; the airline industry floundered and AT&T stock fell. The author completed his doctorate in 1994, leaving the university. He will not return.

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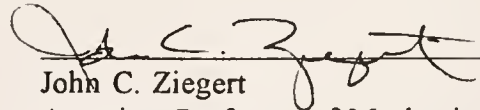
Jim Flusty, Chairman
Graduate Research Professor of
Mechanical Engineering

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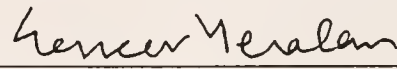
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This dissertation was submitted to the Graduate Faculty of the College of Engineering and to the Graduate School and was accepted as partial fulfillment of the requirements for the degree of Doctor of Philosophy.

December, 1994



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